IN THE UNITED STATES DISTRICT COURT
FOR THE DISTRICT OF DELAWARE

VONAGE HOLDINGS, CORP.,	)	
	)	
Plaintiff,	)	
	)	C.A. No. ______________
v.	)	
	)	
NORTEL NETWORKS, INC. and	)	
NORTEL NETWORKS, LTD.,	)	
	)	
Defendants.	)	

COMPLAINT FOR DECLARATORY JUDGMENT

Plaintiff Vonage Holdings Corp. ("Vonage") states and alleges as follows:

THE PARTIES

1. Plaintiff Vonage is a Delaware corporation, with a principal place of business at 23 Main Street, Holmdel, New Jersey, 07733.

2. Defendant Nortel Networks, Inc. ("NNI") is a Delaware corporation, with, upon information and belief, a principal place of business in Research Triangle, North Carolina.

3. Defendant Nortel Networks, Ltd. ("NNL") is a Canadian corporation, with, upon information and belief, a principal place of business in Ontario, Canada.

4. On information and belief, Defendant NNI is a wholly-owned subsidiary of NNL (hereinafter, collectively "Nortel").

NATURE AND BASIS OF ACTION

5. This is an action for Declaratory Judgment under 28 U.S.C. §§ 2201(a) and 2202, and under the laws of the United States concerning actions related to patents under 28 U.S.C. § 1338(a), arising from an actual controversy between the parties with regard to the invalidity,

unenforceability, and non-infringement of United States Patent Nos. 6,091,808, 6,445,695, and 7,050,861.

JURISDICTION AND VENUE

6. This Court has subject matter jurisdiction in accordance with 28 U.S.C. §§ 2201(a) and 2202, 28 U.S.C. § 1331, and 28 U.S.C. § 1338(a).

7. Venue in this District is proper under 28 U.S.C. § 1391(b), (c), and (d), as both Vonage and NNI are incorporated in Delaware, and NNL is an alien corporation.

THE PATENTS-IN-SUIT

8. Upon information and belief, NNL is the owner of United States Patent No. 6,091,808 ("'808 Patent"), issued on July 18, 2000, for "methods of and apparatus for providing telephone call control and information." A copy of the '808 Patent is attached hereto as Exhibit A.

9. Upon information and belief, NNL is the owner of United States Patent No. 6,445,695 ("'695 Patent"), issued on September 3, 2002, for a "system and method for supporting communications services on behalf of a communications device which cannot provide those services itself." A copy of the '695 Patent is attached hereto as Exhibit B.

10. Upon information and belief, NNL is the owner of United States Patent No. 7,050,861 ("'861 Patent"), issued on May 23, 2006 for "controlling a destination terminal from an originating terminal." A copy of the '861 Patent is attached hereto as Exhibit C.

BACKGROUND FACTS

11. Vonage has been and is currently engaged in the distribution and sale of Internet telephony services and products.

12. Upon information and belief, Nortel has been and is currently engaged in the manufacture and sale of Voice over Internet Protocol ("VoIP") equipment and services.

13. NNI and Vonage are currently both parties to another action in the Northern District of Texas (No. 4-04-CV-548-Y, consolidated with 4-05-CV-224-Y) (hereinafter "Texas Action"), which was commenced on July 27, 2004.[1]

14. In the Texas Action, Vonage has asserted infringement of three patents related to voice compression, U.S. Patent Nos. 4,782,485, 5,018,136, and 5,444,707 (the "Texas Patents in Suit"), against six defendants, including NNI.

15. In the Texas Action, NNI has sought declaratory relief against Vonage, seeking a declaration of non-infringement and invalidity of the Texas Patents in Suit.

16. On July 17, 2007, NNI moved for leave to amend its pleadings in the Texas Action in order to add new infringement claims under the unrelated, and previously unasserted, '808, '695, and '861 Patents that are at issue in this complaint (the "Delaware Patents in Suit"). In the Texas Action, NNI attempted to assert the Delaware Patents in Suit, despite the fact that NNL is listed as those patents' actual owner, according to the U.S. Patent and Trademark Office assignment records, and NNL is not a party to the Texas Action.

17. On August 7, 2007, Vonage opposed NNI's Motion for Leave to Amend under Federal Rules of Civil Procedure 15(a) and 16(b) because it was untimely, raised unrelated claims, and would unduly delay resolution of the Texas Action. NNI has not yet filed its Reply and the Texas Court has not ruled on this motion.

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[1] NNL is not a party to the Texas Action.

NORTEL'S THREATS OF INFRINGEMENT

18. Through its actions and conduct, including but not limited to the allegations of paragraphs 16 and 17, Nortel has put Vonage in the position of continuing to engage in behavior that Nortel asserts is infringing its patents, *i.e.*, continuing to manufacture or distribute its VoIP services and equipment in their current form.

19. Further, through its actions and conduct, Nortel has demonstrated that a substantial controversy exists between parties – Vonage and Nortel – having adverse legal interests, of sufficient immediacy and reality to warrant the issuance of a declaratory judgment.

COUNT I:
NON-INFRINGEMENT OF THE '808 PATENT

20. Vonage incorporates the allegations of paragraphs 1-19 as if fully set forth herein.

21. Nortel has alleged and claimed that Vonage infringes the '808 patent.

22. Vonage does not infringe any valid claim of the '808 patent and has not induced or contributed to the infringement of any valid claim of the '808 patent by another.

23. Vonage is entitled to a judicial declaration that it does not infringe the '808 patent.

COUNT II:
INVALIDITY OF THE '808 PATENT

24. Vonage incorporates the allegations of paragraphs 1-23 as if fully set forth herein.

25. On information and belief, the '808 patent is invalid for failure to meet the conditions of patentability set forth in 35 U.S.C. § 102, § 103, and/or § 112.

26. Vonage is entitled to a judicial declaration that the '808 patent is invalid.

COUNT III:
UNENFORCEABILITY OF THE '808 PATENT

27. Vonage incorporates the allegations of paragraphs 1-26 as if fully set forth herein.

28. On information and belief, the '808 patent is unenforceable for one or more of the grounds alleged in paragraphs 1-26 herein and/or due to laches, estoppel, unclean hands, and/or implied license.

29. Vonage is entitled to a judicial declaration that the '808 patent is unenforceable.

COUNT IV:
NON-INFRINGEMENT OF THE '695 PATENT

30. Vonage incorporates the allegations of paragraphs 1-29 as if fully set forth herein.

31. Nortel has alleged and claimed that Vonage infringes the '695 patent.

32. Vonage does not infringe any valid claim of the '695 patent and has not induced or contributed to the infringement of any valid claim of the '695 patent by another.

33. Vonage is entitled to a judicial declaration that it does not infringe the '695 patent.

COUNT V:
INVALIDITY OF THE '695 PATENT

34. Vonage incorporates the allegations of paragraphs 1-33 as if fully set forth herein.

35. On information and belief, the '695 patent is invalid for failure to meet the conditions of patentability set forth in 35 U.S.C. § 102, § 103, and/or § 112.

36. Vonage is entitled to a judicial declaration that the '695 patent is invalid.

COUNT VI:
UNENFORCEABILITY OF THE '695 PATENT

37. Vonage incorporates the allegations of paragraphs 1-36 as if fully set forth herein.

38. On information and belief, the '695 patent is unenforceable for one or more of the grounds alleged in paragraphs 1-36 herein and/or due to laches, unclean hands, estoppel and/or implied license.

39. Vonage is entitled to a judicial declaration that the '695 patent is unenforceable.

COUNT VII:
NON-INFRINGEMENT OF THE '861 PATENT

40. Vonage incorporates the allegations of paragraphs 1-39 as if fully set forth herein.

41. Nortel has alleged and claimed that Vonage infringes the '861 patent.

42. Vonage does not infringe any valid claim of the '861 patent and has not induced or contributed to the infringement of any valid claim of the '861 patent by another.

43. Vonage is entitled to a judicial declaration that it does not infringe the '861 patent.

COUNT VIII:
INVALIDITY OF THE '861 PATENT

44. Vonage incorporates the allegations of paragraphs 1-43 as if fully set forth herein.

45. On information and belief, the '861 patent is invalid for failure to meet the conditions of patentability set forth in 35 U.S.C. § 102, § 103, and/or § 112.

46. Vonage is entitled to a judicial declaration that the '861 patent is invalid.

COUNT IX:
UNENFORCEABILITY OF THE '861 PATENT

47. Vonage incorporates the allegations of paragraphs 1-46 as if fully set forth herein.

48. On information and belief, the '861 patent is unenforceable for one or more of the grounds alleged in paragraphs 1-45 herein and/or due to laches, estoppel and/or implied license.

49. Vonage is entitled to a judicial declaration that the '861 patent is unenforceable.

JURY DEMAND

In accordance with Fed. R. Civ. P. 38(b), Vonage hereby respectfully requests a trial by jury on all issues so triable.

PRAYER FOR RELIEF

WHEREFORE, Vonage respectfully requests the following relief:

A. A declaratory judgment that the '808 patent is invalid, void and/or unenforceable;

B. A declaratory judgment that Vonage has not infringed the '808 patent;

C. A declaratory judgment that the '695 patent is invalid, void and/or unenforceable;

D. A declaratory judgment that Vonage has not infringed the '695 patent;

E. A declaratory judgment that the '861 patent is invalid, void and/or unenforceable;

F. A declaratory judgment that Vonage has not infringed the '861 patent;

G. A preliminary and permanent injunction enjoining Nortel from in any alleging or asserting patent infringement against Vonage, any of Vonage's current or prospective customers, distributors, dealers, licensees, agents, servants, or employees based on the '808, '695, or '861 Patents.

H. A declaratory judgment that Vonage has not violated any other rights of Nortel.

I. An order awarding Vonage its reasonable costs and attorneys' fees, in accordance with 35 U.S.C. § 285 and other applicable law.

J. An order awarding such other and further relief as the Court deems just and equitable.

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Dated: August 17, 2007
183321.1

EXHIBIT A

US006091808A

United States Patent [19]

Wood et al.

[11] Patent Number: 6,091,808

[45] Date of Patent: Jul. 18, 2000

[54] **METHODS OF AND APPARATUS FOR PROVIDING TELEPHONE CALL CONTROL AND INFORMATION**

[75] Inventors: **Timothy John Wood**; **John C. Anderson**, both of Nepean; **Shirley-Ann Milaknis**, Kanata, all of Canada

[73] Assignee: **Nortel Networks Corporation**, Montreal, Canada

[21] Appl. No.: **08/730,856**

[22] Filed: **Oct. 17, 1996**

[51] **Int. Cl.**[7] **H04M 3/42**

[52] **U.S. Cl.** **379/201**; 379/93.23; 379/216; 379/242; 379/355; 370/352

[58] **Field of Search** 379/201, 209, 379/215, 265, 93.35, 216, 202, 203, 204, 205, 93.24, 242, 93.23; 370/352, 355

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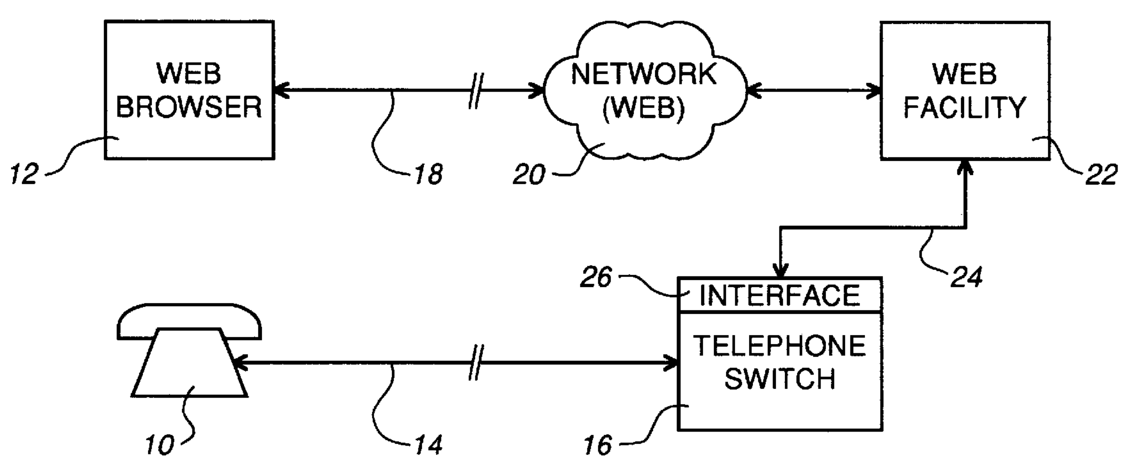
Primary Examiner—Scott Wolinsky
Attorney, Agent, or Firm—R. John Haley

[57] **ABSTRACT**

Telephone call management is provided via a computer network (web) facility which can be remotely accessed by subscribers using web browsers. The web facility includes an information database for storing personal telephone directories and call logs, and a telephone call control system coupled to a telephone switch via a switch-computer interface. Information on calls to and/or from telephone numbers of subscribers is communicated from the switch to the web facility to be stored in the database without requiring the subscribers' browsers to be active. Subscribers can make telephone calls and control telephone communications via the browsers and the web facility. Subscribers do not require any hardware or software in addition to a telephone and web browser.

14 Claims, 2 Drawing Sheets



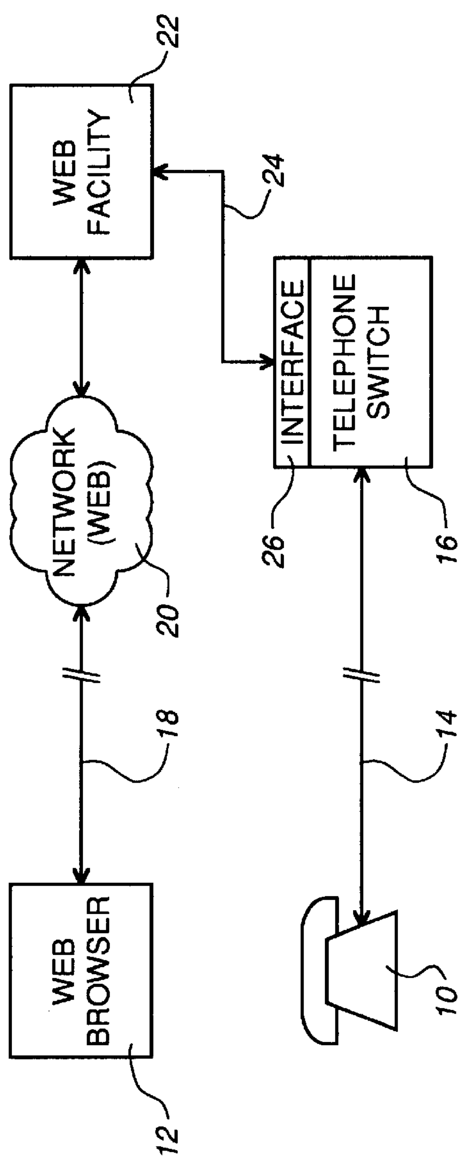

WEB BROWSER
12
18
NETWORK (WEB)
20
WEB FACILITY
22
24
INTERFACE
TELEPHONE SWITCH
26
16
14
10

Fig. 1

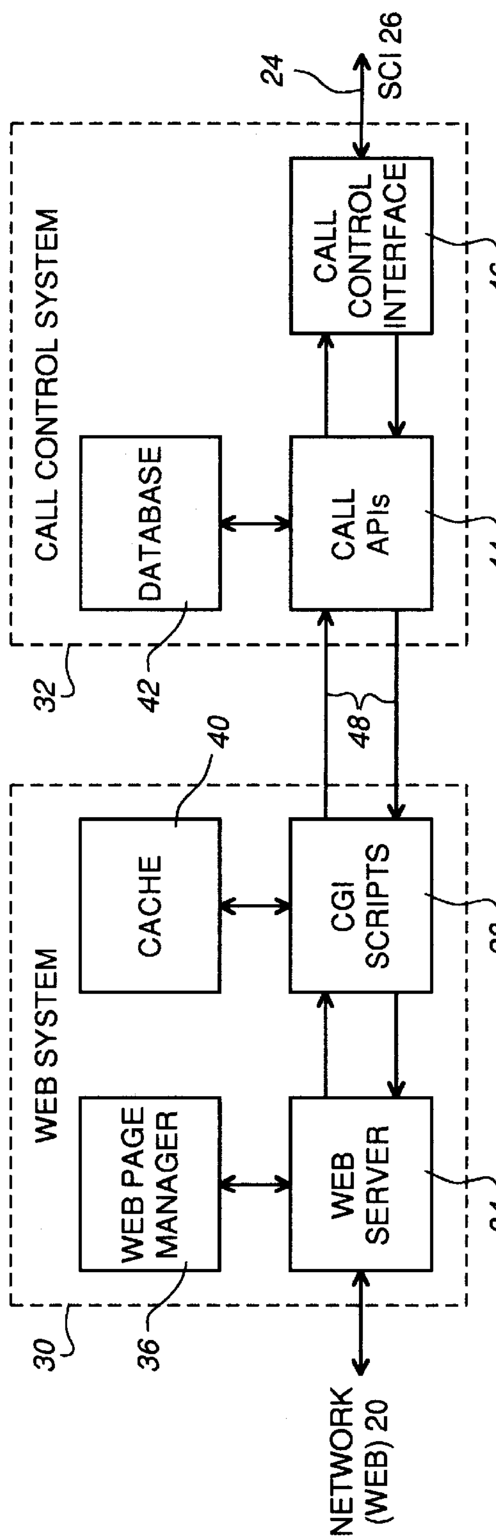

WEB SYSTEM
30
WEB PAGE MANAGER
36
WEB SERVER
34
CACHE
40
CGI SCRIPTS
38
48
CALL CONTROL SYSTEM
32
DATABASE
42
CALL APIs
44
CALL CONTROL INTERFACE
46
24
SCI 26
NETWORK (WEB) 20

Fig. 2

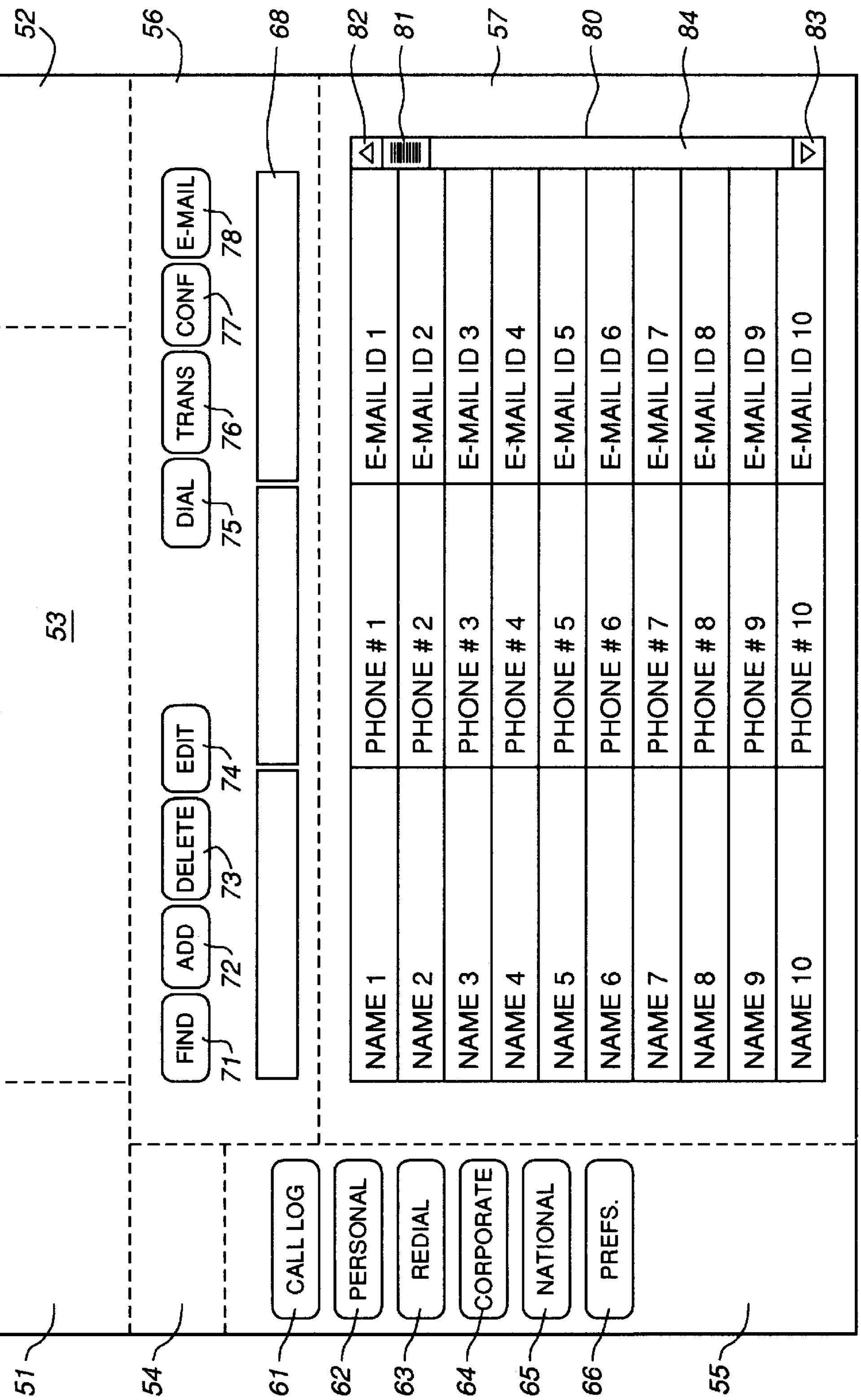

53
FIND
ADD
DELETE
EDIT
DIAL
TRANS
CONF
E-MAIL
NAME 1
PHONE # 1
E-MAIL ID 1
NAME 2
PHONE # 2
E-MAIL ID 2
NAME 3
PHONE # 3
E-MAIL ID 3
NAME 4
PHONE # 4
E-MAIL ID 4
NAME 5
PHONE # 5
E-MAIL ID 5
NAME 6
PHONE # 6
E-MAIL ID 6
NAME 7
PHONE # 7
E-MAIL ID 7
NAME 8
PHONE # 8
E-MAIL ID 8
NAME 9
PHONE # 9
E-MAIL ID 9
NAME 10
PHONE # 10
E-MAIL ID 10
CALL LOG
PERSONAL
REDIAL
CORPORATE
NATIONAL
PREFS.
51
52
54
55
56
57
61
62
63
64
65
66
68
71
72
73
74
75
76
77
78
80
81
82
83
84

Fig. 3

METHODS OF AND APPARATUS FOR PROVIDING TELEPHONE CALL CONTROL AND INFORMATION

This invention relates to methods of and apparatus for providing telephone call control and information.

BACKGROUND OF THE INVENTION

It is well known to provide relatively sophisticated telephone call control and information features using a subscriber's telephone. Some examples of telephone call control features are dialling of stored numbers, redialling of previously dialled numbers, three-way calling, and call forwarding. Examples of telephone information features are calling number display, calling number logs, and call waiting messages. Numerous other examples of call control and information features exist.

Providing such features using the subscriber's telephone involves several disadvantages. For example, the telephone must be capable of providing the required control input and information display functions, so that it becomes a relatively complicated and expensive device. As further call control and information features are developed and become available, the telephone may be unable to accommodate them so that it must be replaced or upgraded to permit use of these further features. Even when the necessary functions are present in the telephone, use of the various functions is not generally simple or intuitive, typically requiring the subscriber to enter various number sequences and/or to interpret relatively cryptic displayed messages. Furthermore, these functions are limited to each individual telephone device, and they must be provided separately for different telephone devices.

Some of these disadvantages have been avoided or reduced by the use of computer-telephone integration (CTI) software which is run on a subscriber's computer in association with telephone control hardware such as a modem or telephone dialler. Such software can facilitate the display of information to, and the input of control information by, the subscriber, and in addition to the features discussed above can facilitate the provision of other features such as telephone directories and voice massaging. However, these CTI arrangements also have several disadvantages. In particular, they require the use of a computer, software, and telephone control hardware by the subscriber, and the computer system must be running continuously to collect information on incoming calls. In addition, such arrangements only provide information at the location at which the system is installed.

More sophisticated arrangements are also known for use with private branch exchange (PBX) and key system telephone networks deployed over a local area network (LAN), with similar disadvantages.

It is also known to provide so-called web call center applications. In this case a subscriber uses a web browser, which for example may be constituted by software running on the subscriber's computer system, to access a computer network such as the international computer network generally referred to as the Internet or World Wide Web, which for brevity is referred to below simply as the web. On browsing a company's web site and desiring to talk with a customer representative of the company, the subscriber can enter his name and telephone number into an HTML (hypertext markup language) page and click on a "submit" button, in response to which the company's telephone system initiates a telephone call from an available representative back to the subscriber. Such call center applications do not provide the telephone call control and information features discussed above.

An object of this invention is to provide improved methods of and apparatus for providing telephone call control and information.

SUMMARY OF THE INVENTION

According to one aspect, this invention provides a method of making a telephone connection comprising the steps of: remotely accessing a computer network facility to produce at the computer network facility a telephone connection message including information identifying calling and called telephone numbers; communicating the telephone connection message from the computer network facility to a telephone switch via a switch-computer interface; and establishing a telephone connection between the calling and called telephone numbers from the switch in response to the telephone connection message.

The step of establishing a telephone connection preferably comprises the step of providing a ringing signal to a telephone identified by the calling telephone number.

Preferably the step of remotely accessing the computer network facility comprises providing telephone number information from the computer network facility for remote display to a subscriber identified by the calling telephone number. The telephone number information can comprise a personal telephone directory of the subscriber, and logged information relating to telephone communications to and/or from the calling telephone number. The step of remotely accessing the computer network facility conveniently comprises operating a web browser.

Thus the invention enables subscribers to control telephone connections, and obtain information from telephone directories and call logs, using a web browser without any need for extra hardware to couple the browser to the telephone. Call logs are maintained without requiring the browsers of the subscribers to be active. In addition, the web or computer network facility can be accessed by each subscriber from any location with web access facilities.

Another aspect of the invention provides a telephone call management system comprising: a computer network facility including a server for communications with telephone subscribers, an information database, and a telephone call control system; a telephone switch including a switch-computer interface; and a communications path between the telephone call control system of the computer network facility and the switch-computer interface of the telephone switch; wherein information relating to telephone communications to and/or from telephone numbers of subscribers is communicated via the communications path from the telephone switch to the computer network facility and being stored in the database, and information for controlling telephone communications is communicated via the communications path from the computer network facility to the telephone switch in response to remote access by subscribers to the information database via the server of the computer network facility.

The invention also provides a method of telephone call management, comprising the steps of: storing personal telephone directories and call logs of telephone subscribers for remote access by the subscribers via a web facility; supplying information, relating to at least some telephone communications associated with the telephone subscribers, from a telephone switch to the web facility; updating the personal telephone directories and call logs of the telephone subscribers in dependence upon information supplied by the subscribers by the remote access via the web facility and the information supplied from the telephone switch; and sup-

plying information from the web facility to the telephone switch, for controlling telephone communications for the subscribers via the telephone switch, in response to the remote access by the subscribers via the web facility.

BRIEF DESCRIPTION OF THE DRAWINGS

The invention will be further understood from the following description with reference to the accompanying drawings, in which:

FIG. **1** is a block diagram schematically illustrating an arrangement in accordance with an embodiment of the invention;

FIG. **2** is a block diagram schematically illustrating one form of a web facility of the arrangement of FIG. **1**; and

FIG. **3** illustrates an example of a web page layout which can be provided in the arrangement of FIGS. **1** and **2**.

DETAILED DESCRIPTION

Referring to FIG. **1**, in an arrangement in accordance with an embodiment of the invention a telephone subscriber has at least one telephone **10** and a web browser **12**. The telephone is coupled via a path **14** to a telephone switch **16**, and the web browser is coupled via a path **18** to a network **20** constituting the web (Internet or World Wide Web). The forms of the telephone **10**, web browser **12**, and paths **14** and **18** are entirely arbitrairy, and these can be known or yet to be devised. The telephone switch **16** can be a central office (C.O.) forming part of the public switched telephone network (PSTN), or a PBX or telephone key system which is coupled to the PSTN in a known manner.

For example, the telephone **10** can be a conventional telephone with pulse or DTMF (dual-tone multi-frequency) dialling, with or without any additional functions for control or information display, coupled to the telephone switch **16** via a twisted wire pair constituting the path **14**. For the purposes of this invention, it is observed that even the dialling function of the telephone **10** is not essential and can be dispensed with (although it would of course be required for conventional use of the telephone **10**). Alternatively, the path **14** could be provided via an ISDN (integrated services digital network) line or any other telephone communications path. With wireless communications, the telephone **10** can be a fixed or mobile telephone.

The path **18** can also be of any known or desired form, for example comprising a wireline or wireless data communications path which may be the same as or separate from the path **14**. Likewise, the form of the web browser **12** is entirely arbitrary. For example it may comprise a personal computer executing browser software in known manner, or a dedicated network browsing device, or a web browsing function integrated within another device such as a video game device or a television receiver or other communications device. Similarly, the functions of the web browser **12** and telephone **10** can be integrated into a single unit, with or without other functions, in any desired manner.

Thus there are numerous ways in which the telephone **10** and web browser **12**, and their paths **14** and **18**, can be implemented, for example including a conventional telephone and personal computer executing browsing software coupled via separate twisted wire pair telephone lines (or via a single telephone line using multiplexed communications) to the telephone switch **16** and web **20**, or an integrated mobile unit combining voice communication and network browsing functions coupled via wireless (e.g. infra-red or radio) communication paths to the web and the PSTN.

The arrangement of FIG. **1** also includes a web facility **22** that is coupled to, and thus can be considered as forming part of, the web **20**. Details of the web facility **22** are described below. The web facility **22** is also coupled via a path **24** to a switch-computer interface (SCI) **26** which forms part of the telephone switch **16**. The path **24** for example comprises a communications path providing X.25 communications between the web facility **22** and the SCI **26**, but it can alternatively comprise any other desired form of communications path, including for example an Ethernet communications path via the network or web **20**.

The SCI **26** is a known facility that is provided by the supplier of the telephone switch **16**. For example, in the event that the telephone switch is a DMS™ telephone switch available from Northern Telecom Limited, then the SCI **26** is constituted by CompuCALL™ facilities also available from Northern Telecom Limited for that switch. Other forms of SCI are available for other telephone switches. The interface **26** can use any of a variety of protocols, such as SCAI (Switch-Computer Access Interface), SPI (Service Programming Interface), or OAP (Open Automated Protocol). In addition, higher level interfaces, such as TSAPI (Telephony Server Application Programming Interface), TAPI (Telephony Application Programming Interface), or JTT (Java Telephony Toolkit) can be implemented in the SCI **26**, or in the CCI **46** described below. In any event, the SCI **26** provides on the path **24** information about telephone calls to telephone lines or directory numbers handled by the switch, and can also control the switch in response to control information supplied via the path **24** to establish calls as described further below.

The web facility **22** provides an interface to the subscriber, via the web **20**, path **18**, and browser **12**, in the form of one or more web pages that enable the subscriber to manage at least some and preferably all telephone functions for the telephone **10**. These functions for example can include all of the functions referred to in the introduction, some of which are further discussed below, as well as other functions which may be desired. To this end, the web facility **22** also communicates call control signals and information relating to these telephone functions with the telephone switch **16** via the path **24** and the SCI **26**.

Accordingly, the web facility **22** constitutes a web server interface for subscriber information and call management functions, and can have any form that enables these functions to be provided and that provides corresponding communications with the telephone switch **16** via the path **24** and SCI 26. FIG. **2** illustrates by way of example one form of the web facility **22**. This form of the web facility **22** comprises two computer systems, shown in FIG. **2** within dashed-line boxes and referred to below as a web system and a call control system **32**. By way of example, the web system **30** may comprise a Windows NT™ computer system and the call control system **32** may comprise a DEC Alpha 2100™ computer system.

The division of the web facility **22** between the two computer systems **30** and **32** is convenient for providing a security firewall between the public network **20** to which the web system **30** is connected and the private data within the computer system **32**, but all of the functions of the web facility **22** could alternatively be provided on a single computer system.

The web system **30** supports a web server **34**, a web page manager **36**, CGI (Communications Gateway Interface) scripts **38**, and a cache (working data storage) **40**. The web system **30** may also support an advertisement server (not

shown). The web server **34** and advertisement server are commercially available software applications which need not be described further here. The web page manager **36** is a software application that manages the presentation of the call management web pages to the subscriber via the web **20**, and that can easily be provided in known manner to provide any desired web page appearance. Purely by way of example and illustration, FIG. **3** shows one possible appearance of a call management web page, and this is further described below. The CGI scripts **38** are software procedures that receive high-level calls from the web server **34** and translate these into lower level operations to be executed in conjunction with the cache **40** and the call control system **32**, with parameters being passed to and from the CGI scripts accordingly.

The call control system **32** supports a database **42**, call APIs (Application Program Interfaces) **44**, and a call control interface **46**. The call control interface **46** is a commercially available product, such as Genesys T-Server™, that provides a network or direct interface via the path **24** to the SCI **26** of the telephone switch **16**. The call APIs **44** communicate with the CGI scripts **38** of the web system **30** via paths **48**, and translate CGI script operations into low level operations comprising calls to and from the call control interface **46** and the database **42**. Thus the CGI scripts **38** and call APIs **44** simply provide successively lower level procedures or software routines for handling calls between the web page manager **36** running on the web server **34**, the call control interface **46**, and the database **42** and cache **40**. The database **42** comprises, for example, a commercially available database manager using SQL (structured query language) in a known manner.

The paths **48** are shown in FIG. **2** for convenience as direct paths between the CGI scripts **38** and the call APIs **44**, but they are preferably constituted by InternetProtocol paths communicating remote procedure calls between these units.

Referring to FIG. **3**, one possible appearance of a call management web page provided by the web facility **22** is illustrated. It is emphasized that this, and the following description of call management functions which can be provided, are given purely by way of example and explanation, and the invention is not in any way limited to these examples or the manner in which they are provided.

As shown by dashed lines in FIG. **3**, the web page is divided into frames which are referenced **51** to **57**. The frames **51** and **52** can be used to display logos relating to the call management service and its provider, and the frame **53** can be used to display an advertising banner. The advertising banner can be provided by an advertisement server on the web system **30** as indicated above or externally of the web facility **22** elsewhere on the web **20**. The manner in which advertising banners are called, provided, and displayed is well known in the art and need not be described here.

The frame **54** can be used to display data relating to the subscriber, for example his name, telephone number, and e-mail ID (electronic mail identity), when he is logged on, and otherwise to display a message indicating that the subscriber is not logged on. The frame **55** provides a number of function buttons **61** to **66**, each constituting a hypertext tag in known manner, functions of which can be as described below.

The frame **56** provides editing windows **68** and buttons **71** to **78**, each constituting a hypertext tag, of which the buttons **71** to **74** provide directory functions and the buttons **75** to **78** provide communication functions as described below. The frame **57** provides contents dependent upon the functions selected by the buttons **61** to **66**, as further described below.

On initially accessing the web facility **22**, the web page manager **36** produces the web page for example with logos in the frames **51** and **52**, an advertising banner obtained from the advertisement server in the frame **53**, and with the frame **57** presenting options (e.g. function buttons and/or editing windows) to permit the subscriber to register or log in. On logging in, the web page manager **36** communicates via the functions **38** and **44** to retrieve data for the subscriber from the database **44** and store this data in the cache **40** for convenient and rapid access. This data can include subscriber information which the page manager **36** then displays in the frame **54** as indicated above, preferences previously stored for the subscriber, and personal directories and call data as discussed further below. The page manager **36** then can also present the frames **55** and **56** for example as shown in FIG. **3**, with the frame **57** being blank or containing any desired information.

On clicking the button **62** labelled PERSONAL, via the function **38** the web page manager **36** accesses a personal directory of the subscriber and displays this in a conventional scrolling window **80** within the frame **57**. For example as shown in FIG. **3** each entry in the personal directory can have name, telephone number, and e-mail ID fields which are displayed in the window **80**. A slider **81**, arrows **82** and **83**, and scroll bar **84** permit the subscriber to scroll through the personal directory records. Clicking on any record causes the fields of that record to be reproduced in the editing windows **68**, where the record can be edited and updated by clicking on the button **74** labelled EDIT. A record identified in the windows **68** can be deleted from the personal directory by clicking on the button labelled DELETE. A desired record can be located by the subscriber entering search criteria in the windows **68** and clicking on the button **71** labelled FIND, and new records can be created in the personal directory from the windows **68** by the subscriber clicking on the button **72** labelled ADD. In this manner, via the web page manager **36** and the CGI scripts **38**, the subscriber can set up and maintain the personal directory in the cache **40**. Updating of the database **42** from the cache **40** can be carried out as desired in the background in a known manner.

The above functions of the buttons **71** to **74** do not involve communications via the call control interface **46**. In contrast, the buttons **75** to **77** invoke communications functions which typically involve communications with the telephone switch **16** via the call control interface **46**. For example, clicking on the button **75** labelled DIAL triggers the telephone switch **16** to set up a telephone connection between the subscriber's telephone **10** and a telephone directory number in the windows **68**. This number can be entered and optionally edited by the subscriber by typing at the network browser **12**, selected from the personal directory by clicking on a record in the window **80** as described above, or provided in another manner for example as described further below.

On clicking the DIAL button **75**, the web page manager **36** communicates a message, containing a dial request, a calling telephone number CN of the subscriber (as displayed in the frame **54**), and a called telephone number DN from the windows **68**, via the functions **38** and **44** to the call control interface **46**, via which this message is forwarded via the path **44** and SCI **46** to the telephone switch **16**. The switch **16** checks validity of the telephone numbers and that the subscriber's telephone **10** (calling telephone number CN) is on-hook, and provides a (possibly distinctive) ringing signal to the telephone **10**. The subscriber, expecting this ring signal, takes his telephone **10** off-hook, and this is detected

by the telephone switch **16** in conventional manner, in response to which the switch **16** sets up the desired telephone connection to the called number DN in the same manner as if the number DN had been dialled by the subscriber at the telephone **10**. Error and/or status messages can be communicated from the telephone switch **16** via the SCI **46**, path **44**, and functions **46**, **44**, and **38** to the web page manager **36**, and displayed on the web page, as desired and appropriate.

It can be appreciated that, in the manner described above, the subscriber is able to instigate a telephone call to a desired number through his access to the web page, and not by dialling at the telephone **10**.

In a corresponding manner, the subscriber can transfer an existing telephone call at his telephone **10** to another called number DN in the windows **68** by clicking the button **76** labelled TRANS. The web page manager again communicates the numbers CN and DN, with a call transfer request, to the telephone switch **16** via the functions **38**, **44**, **46**, and **26**, in response to which the switch **16** transfers the call from the telephone **10** (CN) to the called number (DN) and provides error and/or status messages to the web page manager **36** accordingly. Likewise, the subscriber can establish a conference connection to add another called number DN from the windows **68** to an existing telephone call at his telephone **10** by clicking the button **77** labelled CONF. The web page manager again communicates the numbers CN and DN, with a conference request, to the telephone switch **16** via the functions **38**, **44**, **46**, and **26**, in response to which the switch **16** establishes a conference connection of the call involving the telephone **10** (CN) with the additional called number (DN, again providing en-or and/or status messages to the web page manager **36** accordingly.

In response to clicking on the button **78** labelled E-MAIL, the web page manager **36** creates in known manner a window for the subscriber to enter an e-mail message to an e-mail ID from the window **68** or entered by the subscriber in the e-mail window, this being transmitted in known manner via the web **20**. In this manner, electronic mail communications can also be established by the subscriber using the same web interface as for telephone voice communications. Other communications facilities, for example voice mail messages, and other telephony functions, can be similarly provided in analogous manner to the specific examples given above.

For telephone calls incoming to the telephone **10** via the telephone switch **16**, the SCI **26** provides to the web facility **22** information messages containing for example the called and calling numbers, and the date and time of the call. This information is entered by the call APIs **44** into a call log for the respective subscriber in the database **42** via the functions **46** and **44**. This takes place whether or not the subscriber's web browser **12** is active, so that the call log is not dependent on any activity of the subscriber. On log-in to the web page, the call log is supplied to the cache **40** as described above and is available to the subscriber. The subscriber can click the button **61** labelled CALL LOG, in response to which the web page manager **36** displays the call log in a scrolling window in the frame **57** in place of the personal directory. Each record in the call log can for example include a field containing the calling telephone number (e.g. as in the personal directory described above) supplied from the telephone switch **16** via the SCI **26**, a field for a name which can be optionally provided either similarly by the telephone switch **16** by look-up from the calling telephone number, using the subscriber's (i.e. the called number's) personal directory via the database **42** or using other directory facilities such as a corporate directory as discussed below, and a field for the date and time of the call. Other fields, for example for the duration and status (e.g. answered or not) of the call provided by the SCI **26**, and an associated e-mail address as described above and also provided by the database or directory lookup, can also be provided in the call log as desired.

In a similar manner to that described above for the personal directory, the subscriber can scroll through the call log, click on any record to reproduce it in the windows **68**, click on the ADD button **72** to add a corresponding record to the subscriber's personal directory, click on the DIAL button **75** to establish a return call to the calling number, etc.

For alerting the subscriber to an incoming telephone call, a ringing signal is supplied to the telephone **10** in conventional manner. In addition, if the subscriber's web browser (or a sub-set of this such as a Java applet) is active, then the web page manager **36** is supplied with information about the call (e.g. calling number, name, etc. as provided for the call log as described above) via the functions **44** and **38** and provides an informative alert to the subscriber's web page (or applet window). This obviates the need for processing and display facilities in the telephone **10** to provide call information.

Correspondingly, the web facility maintains a called number log, of numbers called by the subscriber. Conveniently this can be similar to the call log described above, and for calls established by the subscriber using the web facility can use information supplied from the web page manager **36** and/or information supplied by the switch **16** via the SCI **26** as described above for incoming telephone calls to the subscriber. The latter information can also be used to maintain this called number log even for calls made in conventional manner from the telephone **10** without use of the web facility **22**, so that the subscriber's web browser **12** does not need to be active for this called log to be maintained. In the same manner as described above for the call log, the subscriber can click on the button **63** labelled REDIAL to display the called number log in a window in the frame **57**, and again the windows **68** and buttons **71** to **78** in the frame **56** can be used by the subscriber to maintain the called number log and, using the DIAL button **75**, to redial previously called numbers.

A directory of employees of a corporation can be maintained by the web facility **22**, for example as part of the database **42**, and can be used as described above to determine names and other information corresponding to supplied telephone numbers. In addition, such a directory can be used generally by the subscriber using the web facility **22**. In response to the subscriber clicking the button **64** labelled CORPORATE, the web page manager **36** in this case presents in the frame **57** a corporate directory search window in which the subscriber can enter search criteria to locate information for anyone in the directory. Such information is then displayed by the web page manager **36** in the frame **57**, for example in a similar manner to the display of the subscriber's personal directory in this frame as described above and illustrated in FIG. **3**. As in that situation and the other situations described above, the subscriber can scroll through the directory information, click on any record to reproduce it in the windows **68**, click on the ADD button **72** to add a corresponding record to the subscriber's personal directory, click on the DIAL button **75** to establish a call to the selected number, etc.

Other directories can also be accessed by the subscriber via the web facility **22**. For example, a national telephone

directory, containing names, addresses, and telephone numbers, maintained elsewhere on the web **20** can be used by the subscriber by clicking on the button **65** labelled NATIONAL. In response to this the web page manager establishes an http (hypertext transfer protocol) link via the web **20** to the web site of the national directory in a known manner, and presents its search window to the subscriber in the frame **57**. As in the case described above, the subscriber can then find information using the national directory, copy and paste it or click on it to reproduce it in the windows **68**, and add the information to the subscriber's personal directory by clicking on the ADD button **72**, dial the number by clicking on the DIAL button **75**, etc. Other directories external to the web facility **22** can be similarly accessed.

The subscriber can also click on the button **66** labelled PREFS., in response to which the web page manager **36** presents in the frame **57** options for the subscriber to set preferences for his use of the web facility **22** in a known manner.

It will be apparent to those of ordinary skill in the art that all of the above functions, and many other functions which may be desired, can be provided in a relatively straightforward manner by simple messages or procedure calls and responses, with appropriate parameters and returned values, between functions of the web facility **22**, and specifically between the web page manager **36** and the call control interface **46**, cache **40**, and database **42** via the CGI scripts **38** and the call APIs **44**. Details of these procedures, parameters, and returned values depend on the particular functions that are provided and the particular manner in which the web facility **22** is implemented. Such details can be routinely determined by persons of ordinary skill and knowledge, and accordingly need not be, and are not, described here.

A number of significant advantages that may not be immediately apparent can be provided by embodiments of the invention. It can be appreciated that a subscriber can use any web browser **12** and any telephone **10** to provide all of the functions which are available via the web facility **22**. Neither of these is required to have any special hardware or software features, beyond very basic capabilities of the telephone **10** and the inherent functioning of the web browser **12**. The subscriber does not need to acquire or maintain any other software or hardware.

The subscriber is able to access his telephone web page on the web facility **22** from any web browser at any location. This enables all of his call management functions to be available to him regardless of where he may be, for example at home, in an office, or travelling using a mobile telephone and web browser. A particular advantage of this is provided if one of the telephone functions available to the subscriber is call forwarding. In this case for example the subscriber can access the web facility **22** from his office, activate via the web facility **22** a call fowarding function which causes the telephone switch **16** to redirect to his office telephone number calls that are directed to his home telephone number, and receive such calls at his office. Conversely, on returning home he can again access the web facility **22** to remove the call forwarding. The web facility **22** controls the telephone switch to effect and remove the call forwarding function in a similar manner to that described above for call transfer, using another button and related procedures to perform these functions.

Numerous other communications functions can be similarly and easily provided in a corresponding manner, and as already stated, the above description is given purely by way of illustration of the functions that may be provided. As can be appreciated, further functions (both known and yet to be developed) can be easily added by the web facility **22**, and these can be made available immediately to the subscriber, possibly on a subscription or pay-per-use basis that enhances revenues to the service provider. Obviously, the same web facility can be used to serve an arbitrary number of subscribers.

Thus although a particular form of the invention has been described above, it can be appreciated that numerous modifications, variations, and adaptations may be made without departing from the scope of the invention as defined in the claims.

What is claimed is:

1. A method of making a telephone connection comprising the steps of:

storing, for access by a computer network facility which is remotely accessible using a web browser, telephone number information relating to a telephone subscriber;

remotely accessing the computer network facility using a web browser for display of said telephone number information to the subscriber;

producing at the computer network facility using the web browser a telephone connection message including information identifying a calling telephone number of the subscriber and a called telephone number;

communicating the telephone connection message from the computer network facility to a telephone switch via a switch-computer interface; and

establishing a telephone connection between the calling and called telephone numbers from the switch in response to the telephone connection message.

2. A method as claimed in claim **1** wherein the step of establishing a telephone connection comprises the step of providing a ringing signal to a telephone identified by the calling telephone number.

3. A method as claimed in claim **1** wherein the telephone number information comprises a personal telephone directory of the subscriber.

4. A method as claimed in claim **1** wherein the telephone number information comprises logged information relating to telephone communications to and/or from the calling telephone number.

5. A method as claimed in claim **1** and further comprising the step of:

communicating information relating to telephone communications to and/or from the calling telephone number from the switch to the computer network facility;

wherein said telephone number information includes said information communicated from the switch.

6. A method as claimed in claim **5** wherein the telephone number information includes personal telephone directory information of the subscriber.

7. A method as claimed in claim **6** wherein the step of establishing a telephone connection comprises the step of providing a ringing signal to a telephone identified by the calling telephone number.

8. A telephone call management system comprising:

a computer network facility including a web server for communications with web browsers of telephone subscribers, an information database for storing telephone number information relating to the subscribers, and a telephone call control system;

a telephone switch including a switch-computer interface; and

a communications path between the telephone call control system of the computer network facility and the switch-computer interface of the telephone switch;

wherein information relating to telephone communications to and/or from telephone numbers of subscribers is communicated via the communications path from the telephone switch to the computer network facility and is stored in the database for the respective subscribers, and information for controlling telephone communications is communicated via the communications path from the computer network facility to the telephone switch in response to remote access by the respective subscribers to the information database via web browsers of the respective subscribers and the web server of the computer network facility.

9. A system as claimed in claim **8** wherein information stored in the database comprises telephone numbers calling and/or called by the telephone subscribers.

10. A system as claimed in claim **9** wherein information stored in the database further comprises personal telephone directories of the telephone subscribers.

11. A system as claimed in claim **8** the information for controlling telephone communications communicated from the computer network facility to the telephone switch comprises information identifying a telephone number of a subscriber remotely accessing the server of the computer network facility and information identifying a telephone connection request and another telephone number associated with the request.

12. A method of telephone call management, comprising the steps of:

storing personal telephone directories and call logs of telephone subscribers for remote access by the subscribers via a web facility;

supplying information, relating to at least some telephone communications associated with the telephone subscribers, from a telephone switch to the web facility;

updating the personal telephone directories and call logs of the telephone subscribers in dependence upon information supplied by the subscribers by the remote access via the web facility and the information supplied from the telephone switch; and

supplying information from the web facility to the telephone switch, for controlling telephone communications for the subscribers via the telephone switch, in response to the remote access by the subscribers via the web facility.

13. A method as claimed in claim **12** wherein the information supplied from the telephone switch to the web facility identifies calling and called telephone numbers of the telephone subscribers.

14. A method as claimed in claim **12** wherein the information supplied from the web facility to the telephone switch identifies subscriber telephone numbers and connection requests identified by the subscribers by the remote access via the web facility.

* * * * *

EXHIBIT B

US006445695B1

(12) **United States Patent**
Christie, IV

(10) **Patent No.: US 6,445,695 B1**
(45) **Date of Patent: Sep. 3, 2002**

(54) **SYSTEM AND METHOD FOR SUPPORTING COMMUNICATIONS SERVICES ON BEHALF OF A COMMUNICATIONS DEVICE WHICH CANNOT PROVIDE THOSE SERVICES ITSELF**

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(73) Assignee: **Nortel Networks Limited**, St. Laurent (CA)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 0 days.

(21) Appl. No.: **09/223,880**

(22) Filed: **Dec. 31, 1998**

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(52) **U.S. Cl.** **370/352**; 370/466

(58) **Field of Search** 370/264, 351–4, 370/389, 395.1, 395.61, 395.5, 419, 463, 465–469, 400–2

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(57) **ABSTRACT**

A system and method for supporting communications services on behalf of a communications device which cannot provide those services itself in a communications network based on functional signaling. A terminal is designed to identify a supporting server/terminal proxy upon initialization. Henceforth, the terminal provides each user input stimulus to the server and responds to stimulus from the server. The server manages the state machine of the terminal, provides supplementary services, and meets protocol requirements for the network interface.

16 Claims, 2 Drawing Sheets

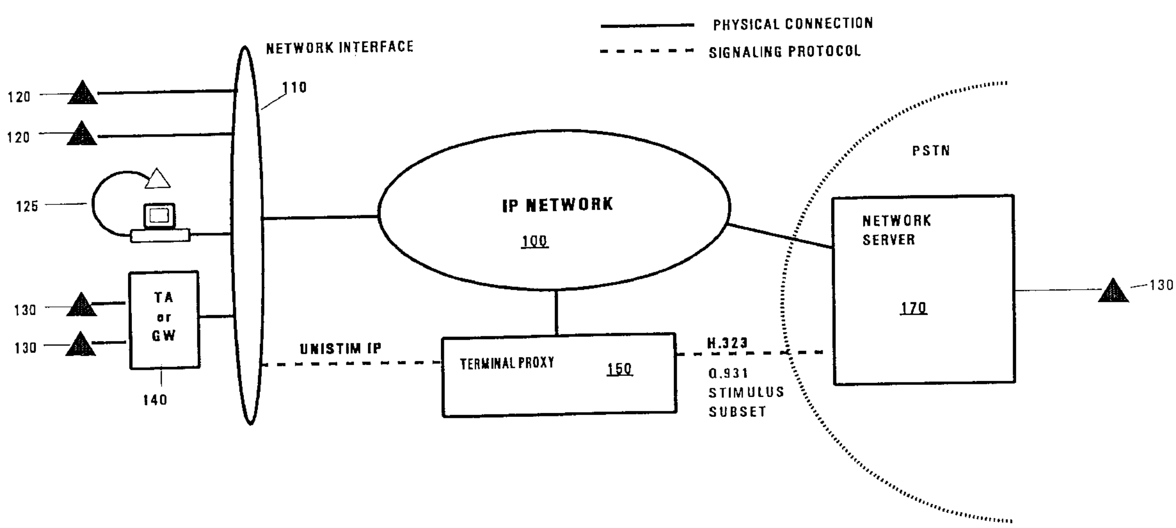


FIG. 1

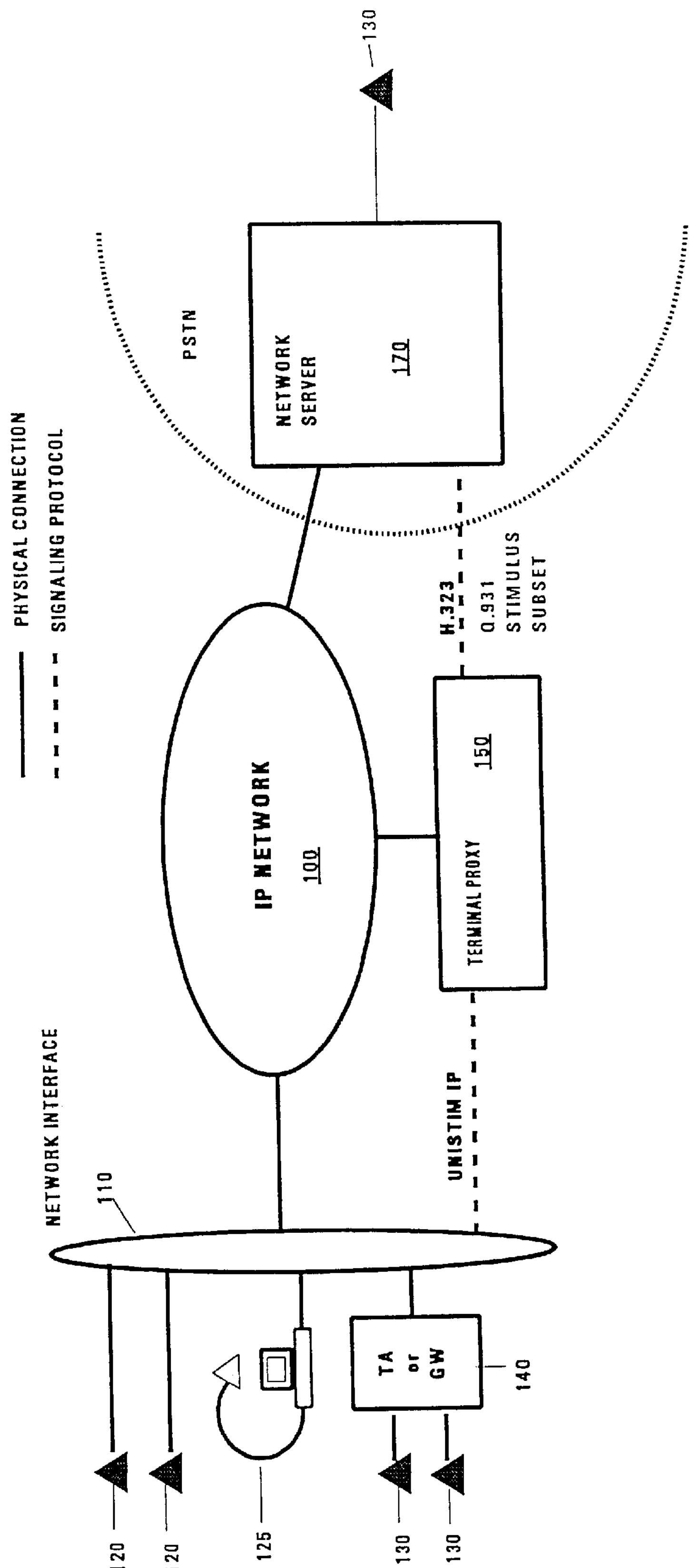
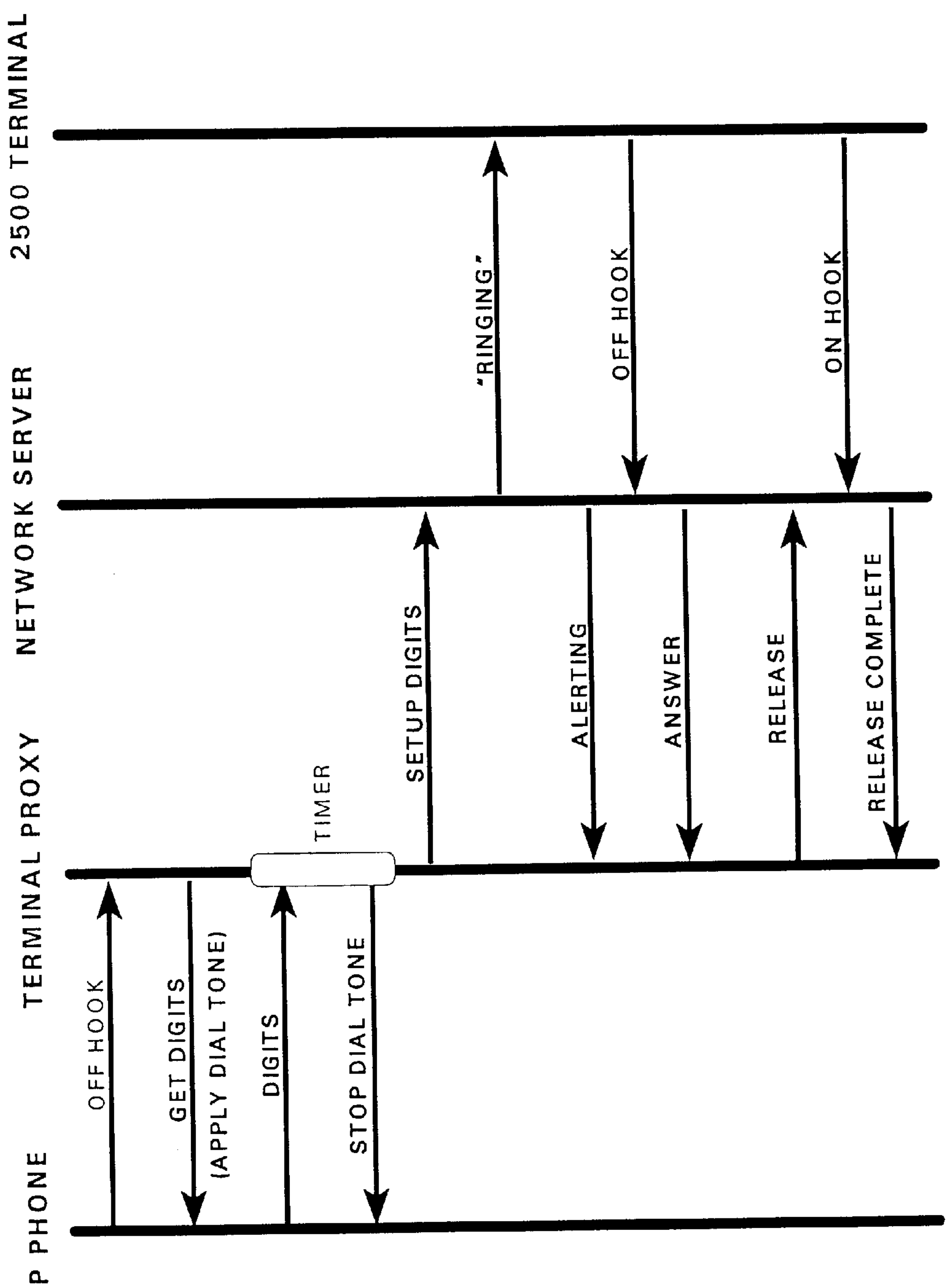

PHYSICAL CONNECTION
SIGNALING PROTOCOL
PSTN
NETWORK SERVER
170
130
IP NETWORK
100
TERMINAL PROXY
150
H.323
Q.931
STIMULUS
SUBSET
UNISTIM IP
NETWORK INTERFACE
110
TA
or
GW
140
120
120
125
130
130

FIG. 2


2500 TERMINAL
NETWORK SERVER
TERMINAL PROXY
IP PHONE
"RINGING"
OFF HOOK
ON HOOK
SETUP DIGITS
ALERTING
ANSWER
RELEASE
RELEASE COMPLETE
TIMER
OFF HOOK
GET DIGITS
(APPLY DIAL TONE)
DIGITS
STOP DIAL TONE

SYSTEM AND METHOD FOR SUPPORTING COMMUNICATIONS SERVICES ON BEHALF OF A COMMUNICATIONS DEVICE WHICH CANNOT PROVIDE THOSE SERVICES ITSELF

TECHNICAL FIELD

The present invention relates to a system and method for supporting communications services on behalf of a communications device which cannot provide those services itself. More specifically, the present invention relates to a system and method for supporting foreign terminals in a communications network.

BACKGROUND AND RELATED ART

In the developing field of IP telephony, there are a multitude of network architecture designs and implementations to consider. The ultimate goal, from a subscriber standpoint, is to provide an IP telephony system that does not require the subscriber to alter the way they perform existing tasks such as placing and receiving calls, and does not affect the set of calling services (caller ID, call forwarding, etc.) that the subscriber may have subscribed to. That is, any new IP telephony system should operate such that subscribers do not have to learn new means for accessing and using services they have always used from their traditional phones should those phones be so called “dumb” terminals.

IP telephony protocols, and there are several under industry-wide consideration, and existing telephony protocols are quite different due to the fact that each handles voice and/or data signals differently. The chief difference is that an IP network partitions a voice and/or data signal into packets and relays the packets over the network from point to point. The packets are then re-arranged at the endpoint for distribution to the consumer in a usable form. Most existing telephony networks are analog in nature meaning signals are not broken into digital packets.

Moreover, IP telephony networks and existing telephony networks must be made compatible with one another in

order to allow an IP telephony subscriber to communicate with a non-IP telephony subscriber and vice-versa. This necessitates network interfaces capable of converting between IP standards and protocols and existing standards and protocols.

Existing telephony networks have an advantage over IP telephony networks in that extensive call processing network hardware is already in place. Moreover, calling services have been developed to operate within existing telephone architecture. Thus, it makes sense to develop an IP telephony system that can utilize to the greatest extent possible the existing network architecture. This primarily entails upgrading certain existing hardware with the previously mentioned network interfaces capable of converting between IP standards and protocols and existing standards and protocols. Another option is to create standalone network interface devices that perform protocol conversion. Such devices would then augment existing architecture. It is to be understood that some of the protocol conversion between IP telephony and existing telephony can be achieved via software. That is, a 100% hardware solution is not required.

As mentioned before, an IP telephony subscriber may want to access the same calling services as existing telephony subscribers. There are a multitude of such services that are currently offered by telephony service providers. These services all require some form of call processing logic to achieve their stated goal. Some services require more extensive processing than others and therefore are network based meaning that the processing logic is performed in a device or node within the telephone network and not by the phone or terminal itself. Other less computationally intensive services may be by the consumer’s own terminal (phone). Additionally, some terminals are “smarter” than others in that they possess greater processing ability and can, therefore, perform certain services themselves as opposed to relying on the network. The table below provides a sampling of calling services and the location at which they may be performed, that is whether the service is network or terminal based. The following table is exemplary, not exhaustive.

TABLE 1

CALLING SERVICE	LOCATION WHERE PERFORMED
CALL FORWARDING NO RESPONSE	NETWORK BASED
CALLING NUMBER DELIVERY/BLOCKING	NETWORK BASED
CLOSED USER GROUP	NETWORK BASED
LOCAL NUMBER PORTABILITY	NETWORK BASED
CONNECTED NUMBER DELIVERY/BLOCKING	NETWORK BASED
EMERGENCY CALL	NETWORK BASED
MALICIOUS CALL TRACE	NETWORK BASED
CALL MONITORING	NETWORK BASED
RELEASE LINE TRUNK	NETWORK BASED OR TERMINAL BASED
ACCOUNT CODE	NETWORK BASED OR TERMINAL BASED
EXTENSION SERVICES	NETWORK BASED OR TERMINAL BASED
MULTIPARTY	NETWORK BASED OR TERMINAL BASED
CALLING NAME DELIVERY	NETWORK BASED OR TERMINAL BASED
800 QUERY	NETWORK BASED OR TERMINAL BASED
IN SERVICE SWITCHING AND RESOURCE FUNCTIONS	NETWORK BASED OR TERMINAL BASED
HOLD AND RETRIEVE	TERMINAL BASED
CALL SCREENING	TERMINAL BASED
CALL TRANSFER	TERMINAL BASED
CALL FORWARDING	TERMINAL BASED
CALL WAITING	TERMINAL BASED
CALL RETURN	TERMINAL BASED
SPEED DIALING	TERMINAL BASED
REPEAT DIALING	TERMINAL BASED

New IP telephony networks define interfaces between a terminal (or client) and a network which necessitate a computationally powerful terminal. The terminal must be capable of managing its call state, providing supplementary services, and managing bearer connections, etc. All of these responsibilities require computational processing capacity (hardware) and software logic. Large numbers of terminals will likely be installed into a network which is expensive in terms of hardware. It is possible to create less computationally intense, less expensive terminals. However, the responsibilities of these scaled back terminals still must be supported. The present invention provides for the delegation of certain terminal responsibilities to a server residing on a network. Thus, use can be made of less expensive, less computationally intensive terminals in an IP telephony network.

The present invention is to be distinguished from IP PBX systems and "black box" devices which allow analog phones to be connected to digital PBX systems. The key difference between a terminal proxy as disclosed by the present invention and an IP PBX controller is that the terminal proxy is not a network call processing engine, rather it is a remote implementation of local call processing. The terminal proxy makes a terminal look like a terminal of another type from the perspective of the IP PBX controller or the central office. The key differentiation with respect to the black boxes which permit analog phones to be connected to digital PBXs is that the black box processes the media and is physically between the PBX and the phone. The terminal proxy, in contrast, is a signaling translator which is not physically between the two machines, only logically so.

SUMMARY OF THE INVENTION

The present invention applies traditional telephony architecture and design to an IP telephony network. To date, thin client (simple terminal) solutions have not been proposed for IP telephony, except in such a way as would eliminate service capabilities or bundle them in a network server such as, for instance, an IP PBX controller.

A terminal according to the present invention is designed to identify its supporting server upon initialization. Henceforth, the terminal provides each user input stimulus to that server (or its backup) and responds to stimulus from the server via a network interface to an IP network using, for instance, the NORTEL proprietary universal stimulus IP protocol (hereinafter "UNISTIM IP"). The server is then responsible for managing the state machine of the client, providing local or terminal specific supplementary services, and meeting protocol requirements for, inter afia, the network interface, SIP, or H.323.

The present invention thus comprises at least one terminal coupled to an IP network and a terminal proxy coupled to the IP network and communicable with the terminals. The terminal proxy communicates with and manages the call processing logic for terminals which cannot perform such tasks for themselves. The present invention further comprises a network interface situated between the terminals and the IP network interface for ensuring that all call control functional signaling between the IP network and the terminals are in a compatible format. The present invention still further comprises a terminal adapter coupled to the IP network via the network interface and supports terminals that do not communicate in an IP protocol. The terminal adapter receives the call control protocol of the terminal and converts it into an IP protocol usable by the IP network.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. **1** is a schematic illustrating one possible network architecture for the present invention; and

FIG. **2** is a message sequence diagram illustrating the call control processing utilized by the terminal proxy methodology of the present invention.

DETAILED DESCRIPTION OF THE INVENTION

The present invention provides a logical means for remote implementation of local call processing that supports calling services and manages call state information and bearer connections in an IP telephony environment on behalf of a terminal which cannot provide those services itself. The premise involves the use of a terminal proxy hosting a client call server residing on the IP network and communicable with "dumb" terminals attached to the IP network. Dumb terminals may take the form of CLASS phones, 2500 sets, IP enabled phones without a high level of computational power, or PCs running IP telephony software. If the terminal happens to be a 2500 set or CLASS phone, then it must be connected to a terminal adapter or gateway mechanism in order to present the proper interface to the IP network. In any event, these dumb terminals will rely on the terminal proxy to perform signaling translation for certain call processing functions. As such, the terminal proxy is not physically between a terminal and an IP PBX. Rather it is logically between the two machines. That is, the terminal proxy makes a terminal appear as another terminal from the perspective of an IP PBX controller.

FIG. **1** illustrates an example of a system architecture for use in the present invention. An IP network **100** is connected to a number of terminal devices through a network interface **110**. The terminal devices illustrated include IP enabled phones **120**, compound terminal(s) **125** comprising more than one physical device such as, for instance, a phone and a computer, and standard 2500 terminals **130**. The 2500 terminals **130** are coupled to the network interface **110** via a terminal adapter/gateway **140**. IP network **100** is also coupled to a terminal proxy **150**. The terminals **120**, **125**, **130** become logical entities to the remainder of the telephony network via terminal proxy **150**.

Additionally, IP network **100** is connected to a network telephony server **170**. Terminals **120**, **125**, **130** communicate with terminal proxy **150** via network interface **110** using a stimulus protocol, most likely UNISTIM IP. UNISTIM IP assumes a call control architecture in which the call control "intelligence" lies outside of the telephone and is thus handled by external call control elements. In the present invention, the call control intelligence is handled either in the terminal itself (rudimentary functions), the terminal proxy **150**, or in network server **170** (complex functionality). An implementation of network server **170** may include a gateway and a PSTN central office switch, PBX, H.323 gatekeeper, or SIP proxy.

Terminal proxy **150** and network server **170** communicate over IP network **100** using the H.323 protocol. The primary H.323 telephony network lelements include a gatekeeper, a terminal, and a gateway. A gatekeeper (GK) is the network entity responsible for IP network address resolution and bandwidth allocation. Terminals provide for real-time two-way communication with one another. The gateway (GW) is the network entity that provides an interface to a non-H.323 network such as, for instance, a public switching telephone network (PSTN). These elements when combined form an H.323 Zone. There may be several gateways in an H.323 zone, each of which provide an interface into some non-H.323 network. The terminals and the gateways are the endpoints of the zone and the gatekeeper manages communications between endpoints of the zone.

To support a plain old telephone service (POTS) terminal **130**, e.g., a 2500 set, in the H.323 IP network **100**, additional network elements are required. POTS phones are connected to IP network **100** via a terminal adapter (TA) **140**. Signaling between POTS terminals and the terminal adapter complies with LATA Switching Systems Generic Requirements (LSSGR). Terminal adapter **140** interacts with the terminal proxy (TP) **150** based on a Stimulus Protocol, such as, for instance SGCP. The stimulus protocol is a messaging system that describes POTS terminal operations. Terminal proxy **150** logically supplies the functional intelligence of an H.323 compliant terminal on behalf of terminal adapter **140** and its supported terminals **130**.

The terminal proxy **150** is a software entity that provides the intelligence and processing for calling services, including basic call processing, on behalf of less capable terminals such as the IP phones **120** and 2500 sets **130** illustrated in FIG. **1**. While the terminal proxy logic could be located on the customer premise, potentially running on the terminal adapter processor, a centralized implementation reduces cost, increases reliability, and facilitates administration. Thus, terminal proxy **150** is a logical remote implementation of local call processing.

One embodiment uses UNISTIM IP as the signaling protocol between terminal adapter **140** and terminal proxy **150** while a stimulus subset of the Q.931 protocol is for communication between terminal proxy **150** and network server **170**.

FIG. **2** illustrates the message sequencing that occurs when an IP phone user initiates a call with a 2500 terminal. An IP Phone subscriber lifts his handset thereby initiating an off hook message which is sent to the terminal proxy. The terminal proxy then sends a return message requesting application of a dial tone and awaits digit entry from the subscriber. The terminal proxy also starts a timer with sufficient time allotted for the subscriber to enter the digits. The subscriber then inputs the digits of the directory number of the terminal that he wishes to call. Upon receipt of the correct number of digits, the terminal proxy sends a message back to the IP phone to stop the dial tone followed by a setup message to the network server serving the IP phone. The setup message informs the network server that a call to the end terminal 2500 set is being made. The network server responds by ringing the desired 2500 Iiterminal while also sending an alerting message back to the terminal proxy. The terminal proxy sends a message requesting a ringback be applied to the IP phone to inform the calling party that the 2500 terminal is ringing. When the called party answers the 2500 terminal, an off hook message is sent to the network server. The network server then forwards an answer message to the terminal proxy. The terminal proxy sends a message request to stop the ringback and open a media channel for cut through audio. The two parties can now engage in a conversation. When the conversation is complete and the IP phone is placed back in its cradle, an on hook message is sent to the terminal proxy signifying the call is over. The terminal proxy responds by sending a release message to the network server. Once the 2500 terminal has “hung up” an on hook message is sent to its network server. The network server responds by sending a release complete message back to the terminal proxy indicating that the calling and called parties have concluded their call and their connection has been terminated. Both terminals are now placed back into a ready state.

The foregoing example illustrates the concept of having the terminal proxy manage the call on behalf of an IP phone. Similar processing would occur had the calling party used a 2500 terminal, or the like, connected to a terminal adapter. In such a case, the terminal adapter is required to interpret the 2500 terminal’s POTS operations.

Call state information, bearer connections, and calling services are handled in the terminal proxy residing on the IP network rather than at the terminal (IP phone). To implement such a system it is necessary for the terminal to identify its supporting server within the IP network upon initialization. This is achieved utilizing standard operations that are well known in the art such as, for instance, DHCP or direct keying into the terminal.

It is to be understood that each of the method or process steps illustrated herein are readily implementable by those of ordinary skill in the art as a computer program product having a medium with a computer program embodied thereon. The computer program product is capable of being loaded and executed on the appropriate computer processing device(s) in order to carry out the method or process steps described. In addition, appropriate computer program code in combination with hardware implements many of the elements of the present invention. This computer code is often stored on storage media. This media can be a diskette, hard disk, CD-ROM, or tape. The media can also be a memory storage device or collection of memory storage devices such as read-only memory (ROM) or random access memory (RAM). Additionally, the computer program code can be transferred to the appropriate hardware over some type of data network.

In the claims, means-plus-function clause are intended to cover the structures described herein as performing the recited function and not only structural equivalents but also equivalent structures. Therefore, it is to be understood that the foregoing is illustrative of the present invention and is not to be construed as limited to the specific embodiments disclosed, and that modifications to the disclosed embodiments, as well as other embodiments, are intended to be included within the scope of the appended claims. The invention is defined by the following claims, with equivalents of the claims to be included therein.

It is to be further understood that the foregoing is illustrative of the present invention and is not to be construed as limited to the specific embodiments disclosed, and that modifications to the disclosed embodiments, as well as other embodiments, are intended to be included within the scope of the appended claims. The invention is defined by the following claims, with equivalents of the claims to be included therein.

What is claimed is:

1. A client proxy signaling system for an IP telephony network comprising:

at least one terminal coupled to an IP network; and

a terminal proxy coupled to said IP network communicable with said at least one terminal for providing logical call processing signaling on behalf of said at least one terminal,

wherein said terminal proxy communicates with and manages the call processing logic for said at least one terminal with respect to the remainder of a telephony network.

2. The system of claim **1** further comprising a terminal adapter coupled to said IP network and said terminal proxy and supporting at least one terminal that does not communicate in an IP protocol, said terminal adapter for receiving the call control protocol of the at least one terminal and converting same into an IP protocol usable by the IP network.

3. The system of claim **2** wherein said IP protocol between the terminal adapter and the terminal proxy is the simple gateway control protocol (SGCP).

4. The system of claim **2** wherein said IP protocol between the terminal proxy and a network server is the H.323 protocol.

5. The system of claim **4** wherein the network server is a PSTN central office switch.

6. The system of claim **4** wherein the network server is a PBX.

7. The system of claim **4** wherein the network server is an H.323 gatekeeper.

8. The system of claim **4** wherein the network server is a SIP proxy.

9. The system of claim **2** wherein said IP protocol between the terminal proxy and a network server is a subset of the Q.931 stimulus protocol.

10. The system of claim **9** wherein the network server is a PSTN central office switch.

11. The system of claim **9** wherein the network server is a PBX.

12. The system of claim **9** wherein the network server is an H.323 gatekeeper.

13. The system of claim **9** wherein the network server is a SIP proxy.

14. A method of providing client proxy signaling in an IP telephony network for a terminal that cannot perform such signaling itself, said method comprising the steps of:

(a) identifying a terminal proxy on an IP network upon initialization of the terminal; and

(b) exchanging call processing messages between the terminal and the terminal proxy such that the terminal proxy manages the call processing logic for the terminal on behalf of the terminal with respect to the remainder of a telephony network.

15. A client proxy signaling system for an IP telephony network comprising:

means for identifying a terminal proxy on an IP network upon initialization of a terminal; and

means for exchanging call processing messages between the terminal and the terminal proxy such that the terminal proxy manages the call processing logic for the terminal on behalf of the terminal with respect to the remainder of a telephony network.

16. A client proxy signaling computer program product for an IP telephony network having a medium with a computer program embodied thereon comprising:

computer program code for identifying a terminal proxy on an IP network upon initialization of a terminal; and

computer program code for exchanging call processing messages between the terminal and the terminal proxy such that the terminal proxy manages the call processing logic for the terminal on behalf of the terminal with respect to the remainder of a telephony network.

* * * * *

EXHIBIT C

US007050861B1

(12) **United States Patent**
Lauzon et al.

(10) **Patent No.: US 7,050,861 B1**
(45) **Date of Patent: May 23, 2006**

(54) **CONTROLLING A DESTINATION TERMINAL FROM AN ORIGINATING TERMINAL**

(75) Inventors: **Eric Lauzon**, Crowthorne (GB); **Bryan J Miller**, Cookham (GB); **Michael O'Doherty**, London (GB)

(73) Assignee: **Nortel Networks Limited**, St. Laurent (CA)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 1052 days.

(21) Appl. No.: **09/606,053**

(22) Filed: **Jun. 28, 2000**

Related U.S. Application Data

(60) Provisional application No. 60/171,777, filed on Dec. 22, 1999, provisional application No. 60/171,801, filed on Dec. 22, 1999.

(51) **Int. Cl.**
G05G 11/00 (2006.01)

(52) **U.S. Cl.** **700/17**; 700/19; 700/65; 700/83; 709/200; 709/208; 709/220; 709/227; 709/228; 379/188; 379/201.01; 379/201.03; 379/201.05; 379/203.01; 379/204.01

(58) **Field of Classification Search** 700/17, 700/65, 83, 19; 345/804; 709/200, 208, 709/220, 227–228; 379/188, 201.01, 201.03, 379/201.05, 203.01, 204.01
See application file for complete search history.

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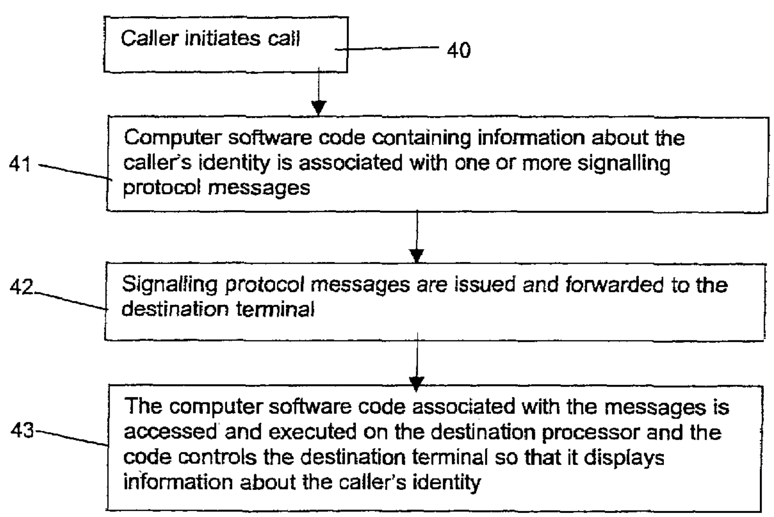
Primary Examiner—Anthony Knight
Assistant Examiner—Ronald D Hartman, Jr.
(74) *Attorney, Agent, or Firm*—Barnes & ThornburgLLP

(57) **ABSTRACT**

A caller associates computer software code with a signalling protocol messages such that when the messages are received at a destination processor the computer software code is executed. For example, the messages may be improved SIP protocol messages with incorporated Java code. By selecting different computer software code for association with the messages, the caller is able to control the destination terminal. For example, to display information about the identity of the caller at the destination terminal; to modify the behaviour of the destination terminal according to the priority of the call; to take into account the configuration of the destination terminal, and to allow users to adjust this configuration from a remote location.

13 Claims, 11 Drawing Sheets



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PRIOR ART

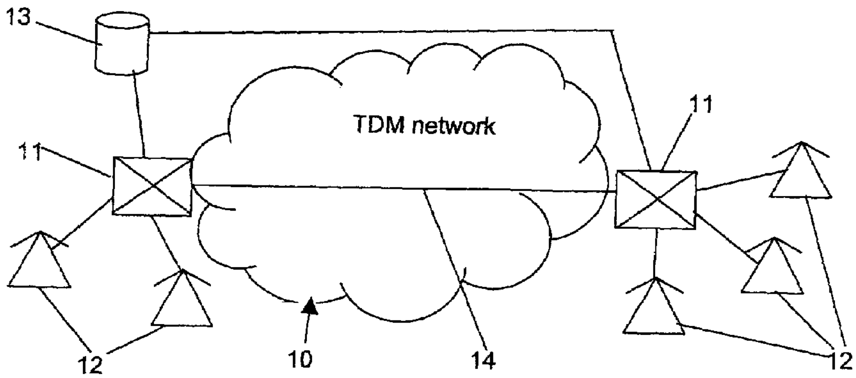


Figure 1

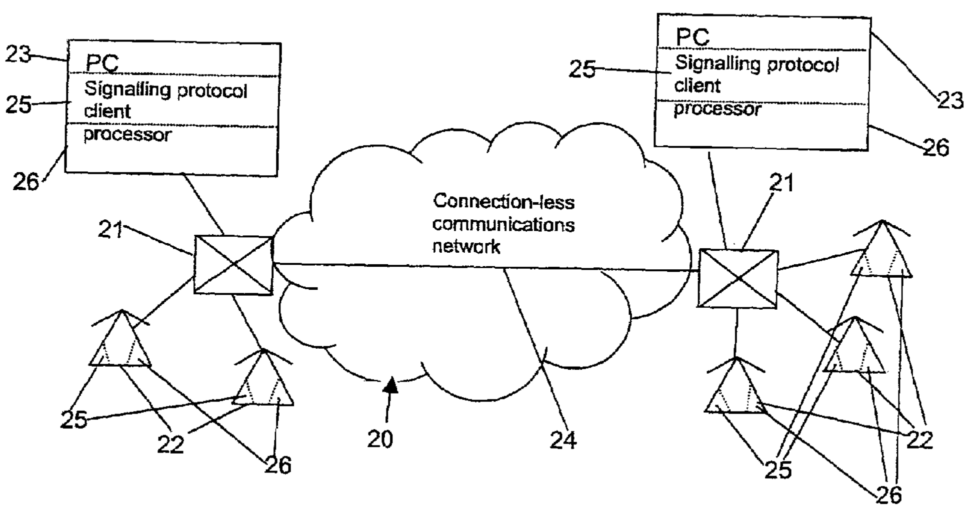


Figure 2

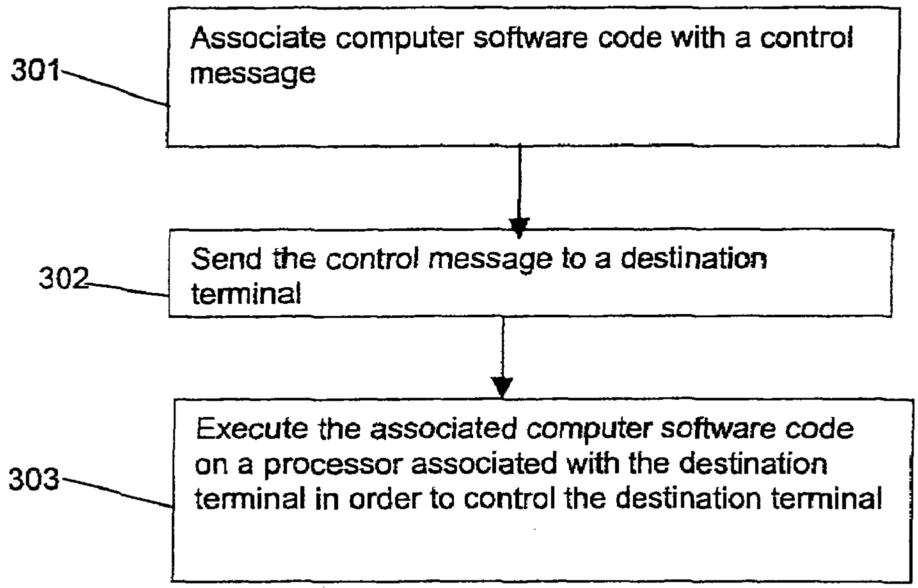


Figure 3

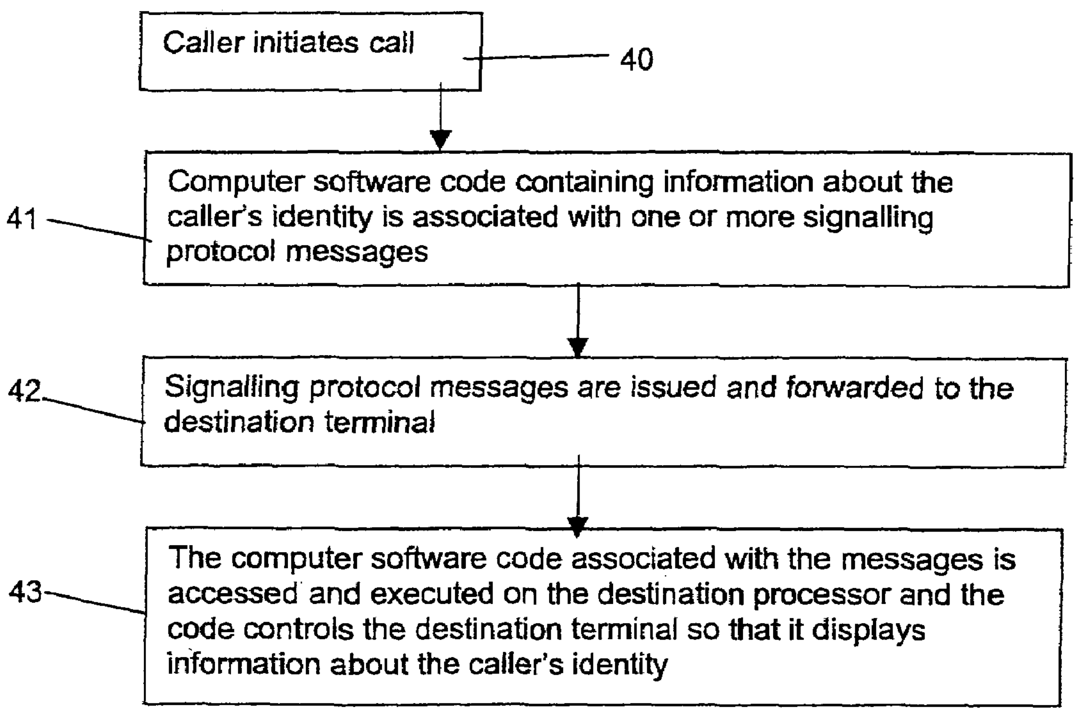


Figure 4

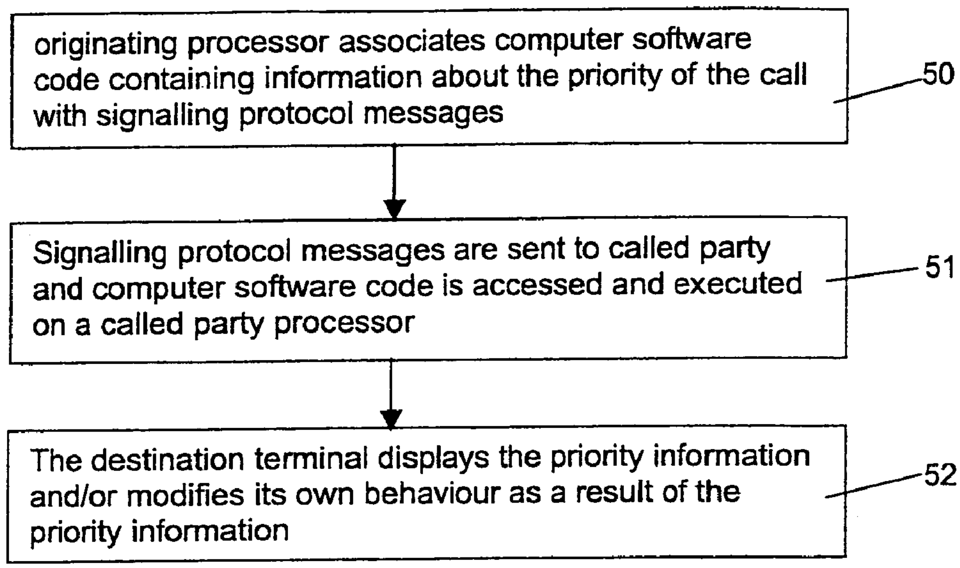


Figure 5

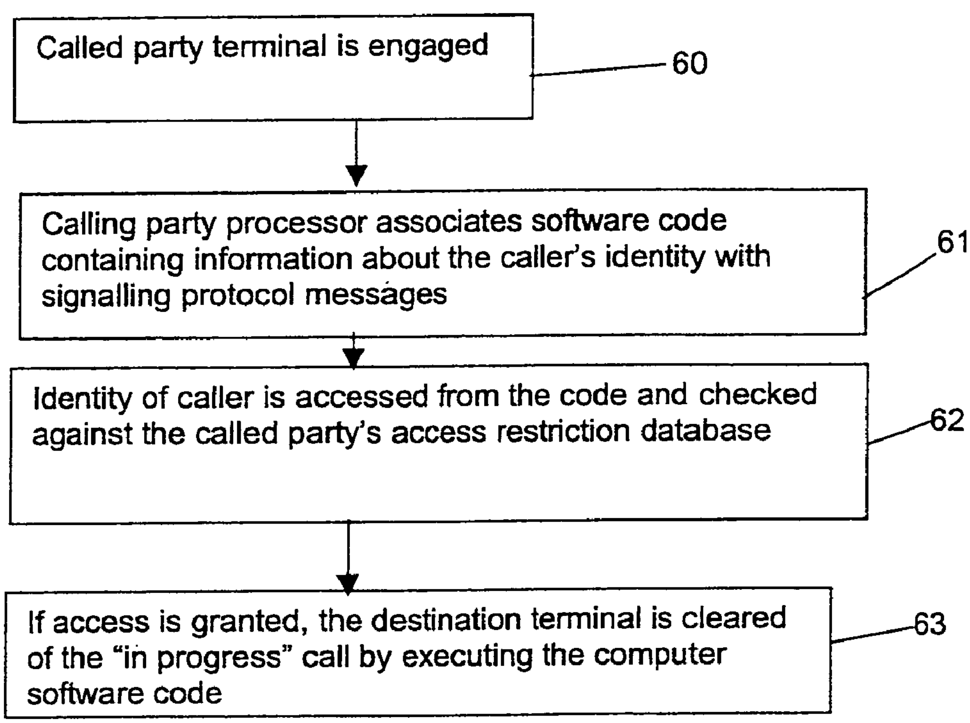


Figure 6

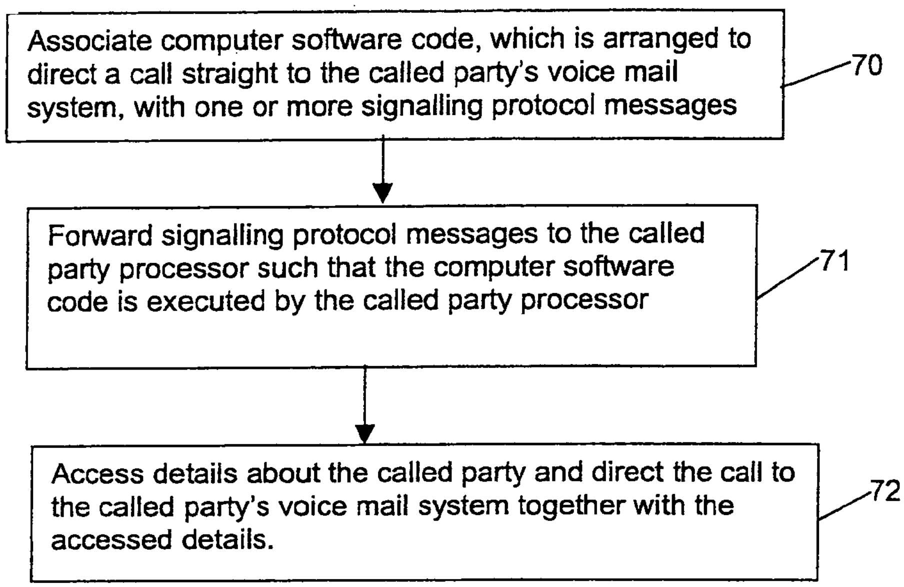


Figure 7

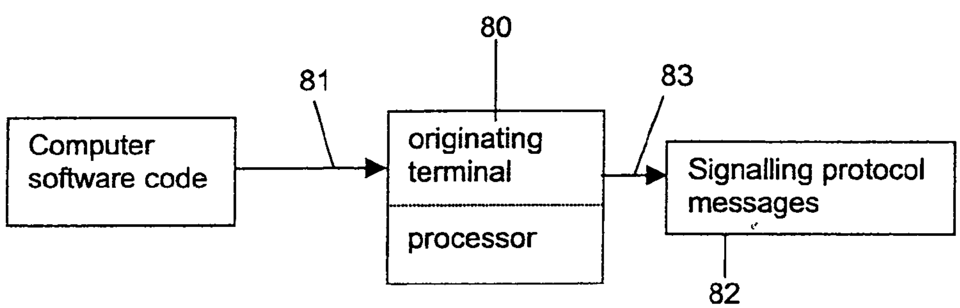


Figure 8

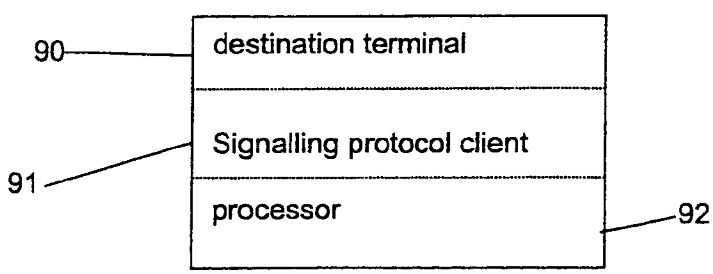


Figure 9

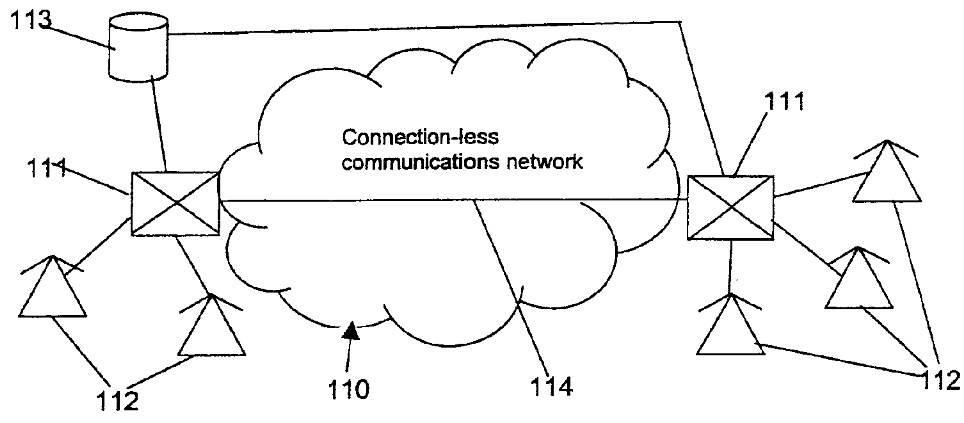


Figure 10

Provide a database comprising information about the identity of a plurality of originating terminals and a caller associated with each originating terminal — 120

↓

Initiating a call from an originating terminal to a destination terminal — 121

↓

Receiving information at the originating terminal about the identity of a caller and forwarding this information from the originating terminal to the database and updating the database with this information — 122

↓

Accessing the identity of the caller associated with the originating terminal from the database and displaying that identity at the destination terminal — 123

Figure 11

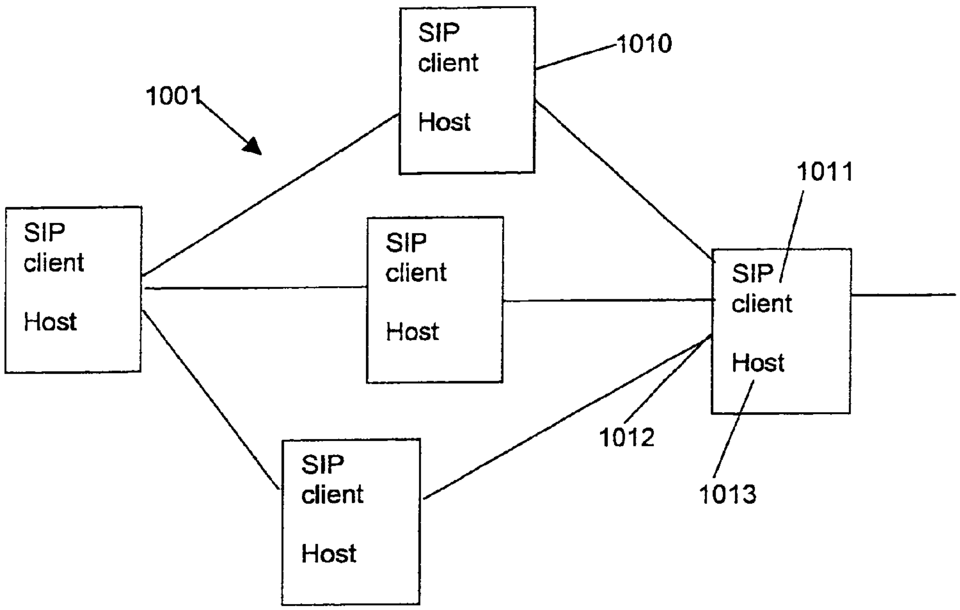


Figure 12

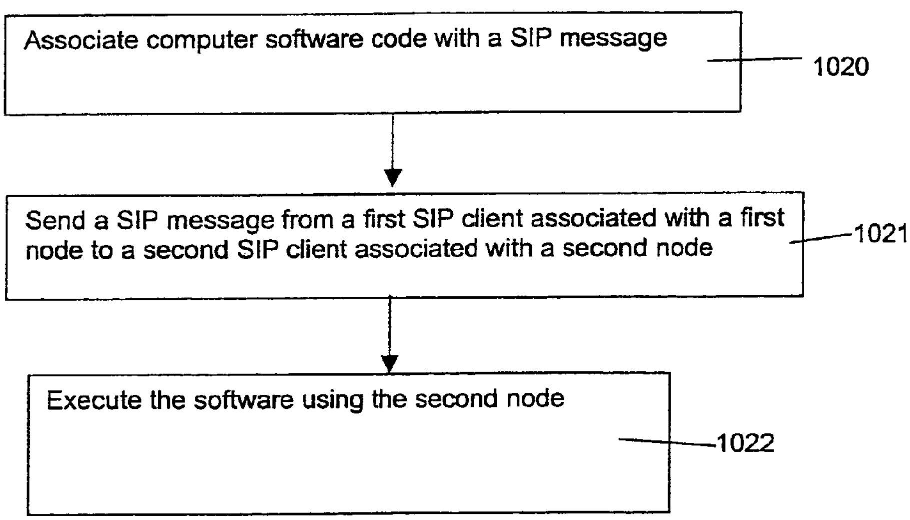
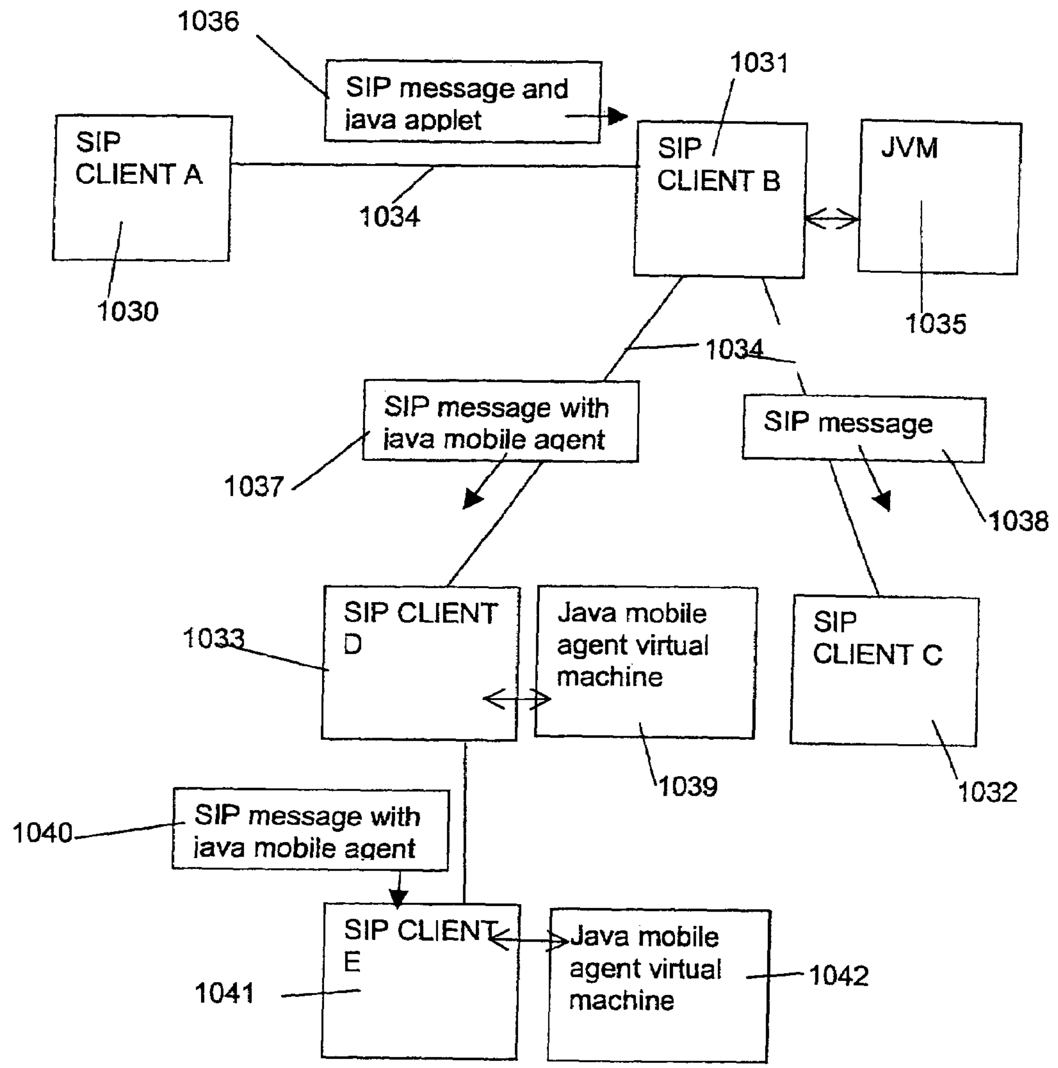


Figure 13



1036
SIP message and java applet
1031
SIP CLIENT A
SIP CLIENT B
JVM
1034
1030
1035
1034
SIP message with java mobile agent
SIP message
1037
1038
SIP CLIENT D
Java mobile agent virtual machine
SIP CLIENT C
1033
1039
1032
1040
SIP message with java mobile agent
SIP CLIENT E
Java mobile agent virtual machine
1041
1042

Figure 14

```
C ->S: INVITE sip:watson@boston.bell-tel.com SIP/2.0
Via: SIP/2.0/UDP kton.bell-tel.com>
From: A. Bell ,sip:a.g.bell@bell-tel.com>
To: T. Watson ,sip:watson@bell-tel.com.
Call-ID: 3298420296@kton.bell.tel.com
Cseq: 1 INVITE
Subject: Mr. Watson, come here.
Content-Type:   multipart/mixed;   boundary=3E4A567F4C8A
(or URL for java applet)
Content-Length: ...
Require: org.ietf.sip.java-enhanced-sip

(Within the message body)

--3E4A567F4C8A
        Content-Type: application/x-sipjava
        Content-Encoding: binary                      1050
        Content-length: xxx
        ...Java applet or Java mobile agent for SIP message
        processing...
--3E4A567F4C8A—
```

Figure 15

```
Request    =  Request-Line
              *( general-header ;
              | request-header
              | entity-header                          1040
              CRLF
              [ message-body ]  ;

general-header  = Accept                     ;
                | Accept-Encoding            ;
                | Accept-Language            ;
                | Call-ID                    ;
                | Contact                    ;
                | CSeq                       ;
                | Date                       ;
                | Encryption                 ;         1041
                | Expires                    ;
                | From                       ;
                | Record-Route               ;
                | Timestamp                  ;
                | To                         ;
                | Via                        ;
     entity-header  = Content-Encoding       ;
1042            | Content-Length             ;
                | Content-Type               ;  This will indicate that
     request-header = Authorization          ;  the content type is a
                | Contact                    ;  java applet or a Java
                | Hide                       ;  Mobile Agent (or the URL
                | Max-Forwards               ;  of a location of either
                | Organization               ;  from where they must be
                | Priority                   ;  retrieved).
1043            | Proxy-Authorization        ;
                | Proxy-Require              ;
                | Route                      ;  This will be used to
                | Require                    ;  indicate that Java-
                | Response-Key               ;  enhanced-SIP must be
                | Subject                    ;  supported to process
                | User-Agent                 ;  this message
response-header = Allow                      ;
                | Proxy-Authenticate         ;
                | Retry-After                ;
                | Server                     ;         1044
                | Unsupported                ;
                | Warning                    ;
                | WWW-Authenticate           ;
```

Figure 16

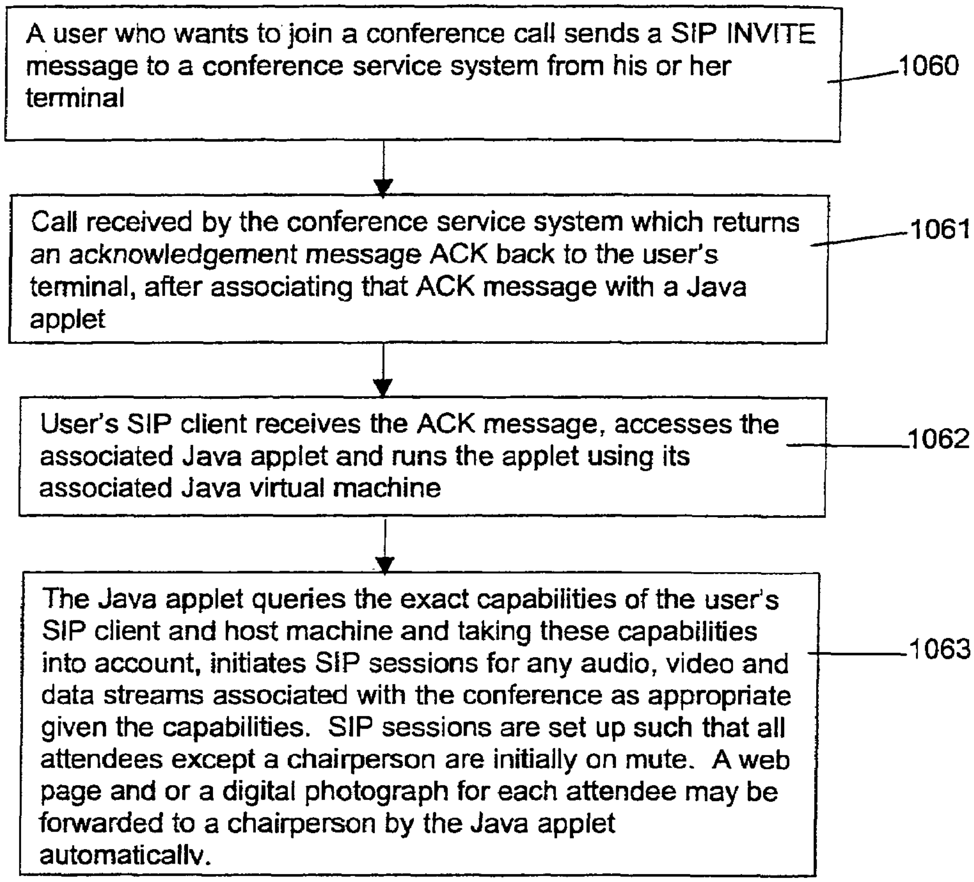


Figure 17

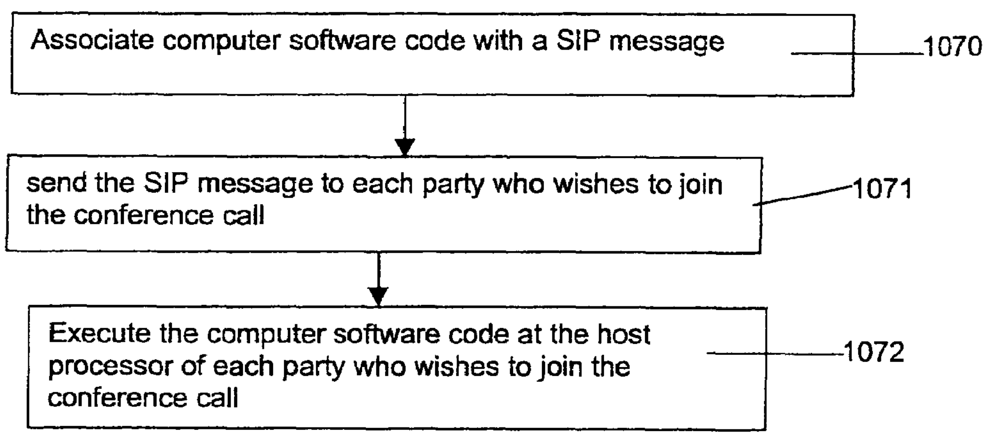


Figure 18

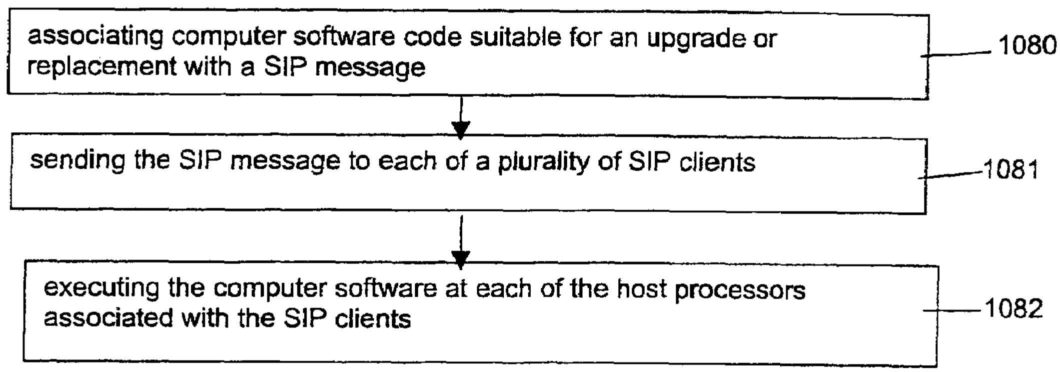


Figure 19

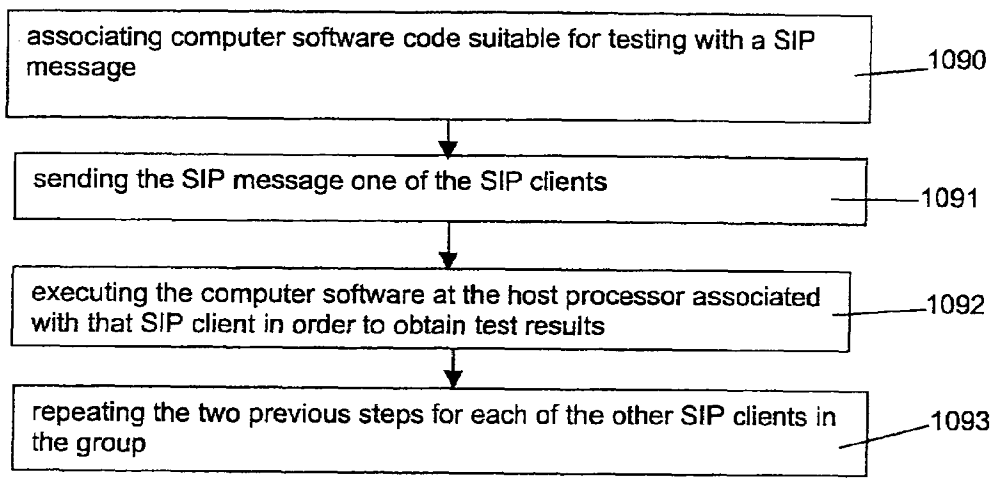


Figure 20

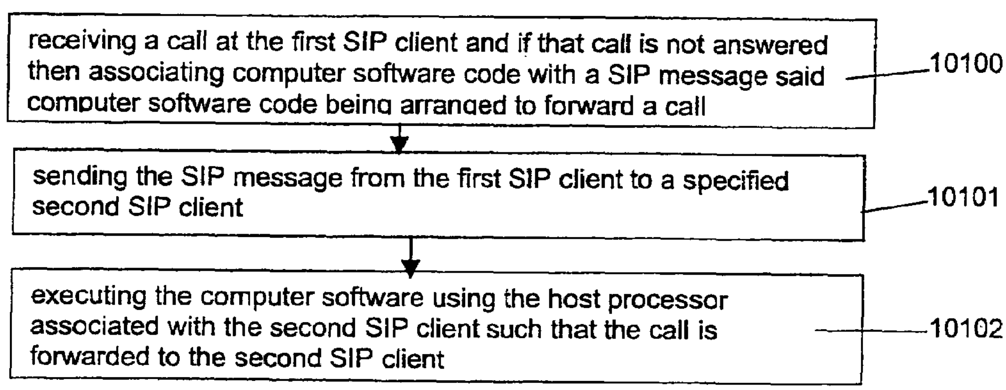


Figure 21

CONTROLLING A DESTINATION TERMINAL FROM AN ORIGINATING TERMINAL

RELATED APPLICATIONS

This application is the non-provisional filing of provisional U.S. Patent Applications 60/171,777 and 60/171,801, both filed on Dec. 22, 1999.

BACKGROUND OF THE INVENTION

1. Field of the Invention

This invention relates to a method of remotely controlling a destination terminal from an originating terminal. The invention is particularly related, but in no way limited to, using improved session initiation protocol (SIP) to enable a caller to control an originating terminal.

2. Description of the Prior Art

The amount of control that an originating terminal has over a destination terminal has been very restricted. For example, when making an extremely urgent call to a busy destination terminal, the caller is unable to free up the busy destination terminal by causing the call that is currently in progress to be dropped. Also, the called party may have particular services set-up on his or her terminal and the calling party is unable to take these into account easily or to modify the set-up services. This is particularly problematic when a caller wishes to adapt his or her call as a result of taking the called party's terminal configuration into account. For example, a user may be accustomed to setting his or her terminal to ring three times before going to voice mail, during times when that user is resting. At other times, suppose that the user sets his or her terminal to ring five times before going to voice mail. The user's family members may wish only to make a call to the user when the user is not resting. However, this is not possible because callers are unable to take into account set-up configurations on the user's terminal.

Similarly, calling parties are unable to easily provide information to the called party and to cause the destination terminal to display or act upon this information. For example, a calling party may wish to provide information about his or her identity to the called party. In the past this has been done by associating each terminal with a particular user. However, this is problematic when users move about and use different terminals. Also, prior art systems which display the caller identity at the destination terminal are fixed systems. That is, the caller is unable to easily change or modify the manner in which the destination terminal displays or acts upon the identity information.

It is accordingly an object of the present invention to provide a method of remotely controlling a destination terminal from an originating terminal, which overcomes or at least mitigates one or more of the problems noted above.

SUMMARY OF THE INVENTION

According to an aspect of the present invention there is provided a method of remotely controlling a destination terminal from an originating terminal said destination terminal having an associated signalling protocol client and an associated processor comprising the steps of:

- associating computer software code with at least one signalling protocol message;
- sending the signalling protocol message to the destination terminal from the originating terminal;

- executing the computer software code using the processor associated with the destination terminal in order that the originating terminal controls the destination terminal.

This provides the advantage that an originating terminal is able to control a destination terminal. For example, to display information about the identity of the caller on the destination terminal or to modify the behaviour of the destination terminal on the basis of priority information provided by the calling party.

According to another aspect of the present invention there is provided an originating terminal arranged to control a destination terminal said originating terminal comprising:—

- an input arranged to access computer software code suitable for controlling said originating terminal;
- a processor arranged to associate said computer software code in use with one or more signalling protocol messages; and
- an output arranged to route said signalling protocol messages to the destination terminal in use.

This provides the advantage that by using such an originating terminal a user is able to control a destination terminal.

According to another aspect of the present invention there is provided a destination terminal comprising:—

- a signalling protocol client arranged to receive one or more signalling protocol messages sent from an originating terminal;
- a processor arranged to access any computer software code associated with received signalling protocol messages in use; and wherein said processor is arranged to execute such accessed computer software code such that the destination terminal is controlled.

This provides the advantage that a destination terminal which can be controlled from a remote location by an originating terminal is provided.

According to another aspect of the present invention there is provided a signal comprising one or more signalling protocol messages which are associated with computer software code. This provides the advantage that the functions of the signalling protocol messages are greatly extended. For example, the signalling protocol messages can be sent from an originating terminal to a destination terminal to control that destination terminal.

According to another aspect of the present invention there is provided a method of displaying information about the identity of a caller at a destination terminal comprising the steps of:

- providing a database comprising information about the identity of a plurality of originating terminals and a caller associated with each originating terminal;
- initiating a call from an originating terminal to a destination terminal;
- receiving information at the originating terminal about the identity of a caller and forwarding this information from the originating terminal to the database and updating the database with this information; and
- accessing the identity of the caller associated with the originating terminal from the database and displaying that identity at the destination terminal.

This provides the advantage that the database of caller identity information is updated prior to use so that the identity information displayed is correct, even if the caller uses different terminals or several users use the same terminal.

Further benefits and advantages of the invention will become apparent from a consideration of the following

detailed description given with reference to the accompanying drawings, which specify and show preferred embodiments of the invention.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. **1** is a schematic diagram of a time division multiplex (TDM) communications network arrangement according to the prior art.

FIG. **2** is a schematic diagram of a connectionless communications network suitable for use with the present invention.

FIG. **3** is a flow diagram of a method of controlling a destination terminal from an originating terminal.

FIG. **4** is a flow diagram of a method of controlling a destination terminal such that information about the identity of the calling party is displayed at the destination terminal.

FIG. **5** is a flow diagram of a method of controlling a destination terminal using information about the priority of the call.

FIG. **6** is a flow diagram of a method of controlling a destination terminal in order to clear an "in progress" call from the destination terminal.

FIG. **7** is a flow diagram of a method for controlling a destination terminal in order that the call is directed straight to a voice mail system.

FIG. **8** is a schematic diagram of an originating terminal.

FIG. **9** is a schematic diagram of a destination terminal.

FIG. **10** is a schematic diagram of a connectionless communications network suitable for use with an embodiment of the present invention

FIG. **11** is a flow diagram of a method of displaying information about the identity of a caller at a destination terminal.

FIG. **12** is a schematic diagram of a communications network which incorporates nodes for implementing an improved SIP protocol.

FIG. **13** is a flow diagram of a method of communicating between two SIP clients using an improved SIP protocol.

FIG. **14** is a schematic diagram of interaction between a plurality of SIP clients according to the improved SIP protocol.

FIG. **15** shows the format of an improved SIP protocol message.

FIG. **16** is an example of an improved SIP protocol INVITE message.

FIG. **17** is a flow diagram of a method of setting up a conference call using a conference call service system.

FIG. **18** is a flow diagram of a method of setting up a conference call.

FIG. **19** shows a method of upgrading or replacing interconnected SIP clients.

FIG. **20** shows a method of testing members of a group of SIP clients.

FIG. **21** shows a method of forwarding a call from a first SIP client to a second SIP client.

DETAILED DESCRIPTION OF THE INVENTION

Embodiments of the present invention are described below by way of example only. These examples represent the best ways of putting the invention into practice that are currently known to the Applicant although they are not the only ways in which this could be achieved.

The term "originating terminal" is used to refer to an apparatus via which a user is able to send communications into a communications network in order to call another party; for example, a telephone handset, a computer terminal or a mobile telephone handset.

The term "destination terminal" is used to refer to an apparatus via which a user is able to receive communications from the communications network in order to be called by another party; for example, a telephone handset, a computer terminal or a mobile telephone handset.

The term "calling party" is used to refer to an entity which sends a communication into a communications network in order to communicate with a called party.

The term "called party" is used to refer to an entity which receives communications from a calling party via a communications network.

The present application is at least in part an extension of Nortel Networks's earlier work described in co-assigned, earlier U.S. patent application Ser. No. 09/520,853, filed on 7 Mar. 2000 (Nortel reference 11790 ID). That patent document describes an improved Session Initiation Protocol (SIP). Using this improved SIP protocol computer software code is associated with SIP messages. These SIP messages are sent to a SIP client which is arranged to execute the software code associated with the SIP messages. The specific description from U.S. patent application Ser. No. 09/520,853 is repeated in Appendix A.

FIG. **1** shows a prior art arrangement in which a plurality of terminals **12** (such as telephone handsets) are connected to a time division multiplex (TDM) communications network (such as a public switched telephone network) via access nodes **11**. A database **13** is also provided which is accessible by each of the access nodes **12**. The database contains pre-specified information about the identity of each terminal **12** (for example, the calling line identifier (CLID)) and the name of a user associated with each terminal. When a caller initiates a call, the name associated with the terminal from which the call is being made is accessed from the database **13** and displayed at the called terminal. In some circumstances this lets the called party know who is calling before the call is answered. However, often one particular terminal is associated with more than one person and in addition, callers are mobile and often use different terminals to make calls. Arrangements like that illustrated in FIG. **1** are not able to deal with these situations and simply display the name of the one user associated with the particular terminal being used, even if a different person is actually using that terminal.

Another prior art arrangement involves storing pre-specified information about the identity of terminals at their associated access nodes **11**. For example, this information comprises the CLID of each terminal **12** which is connected to the access node **11** and the name of a user associated with each of those terminals **12**. When a caller initiates a call, the name associated with the terminal from which the call is being made is sent with the call to the destination terminal. The name information is static and because of this the system is not flexible and cannot take account of the fact that different users use the same originating terminal or that individual users move about and use different terminals.

FIG. **2** illustrates an embodiment of the present invention in which information about the identity of a caller is made available to the called party independently of the particular terminal being used by the caller. A plurality of terminals **22**, **23** are connected to a connectionless communications network **20** such as an internet protocol (IP) communications network via access nodes **21** such as voice over internet protocol (VoIP) gateways. Calls are set-up between two terminals using any suitable signalling protocol such as

session initiation protocol (SIP). The terminals may be for example, personal computer based telephones **23** or conventional telephone handsets **22**. Associated with each terminal is a signalling protocol client **25** which is a computer program that is arranged to control the terminal such that it is able to send and/or receive messages according to the particular signalling protocol being used. This signalling protocol client program **25** may be provided on any suitable computing platform integral with the terminal or accessible by the terminal. As well as the signalling protocol client **25**, a processor **26** is associated with each terminal and the processor **26** is arranged to execute any computer software code that is associated with signalling protocol messages received from callers.

With reference to FIG. **4**, when a caller initiates a call (box **40** of FIG. **4**), computer software code is associated with one or more signalling protocol messages issued by the caller's terminal in order to set-up the call (box **41** of FIG. **4**). This computer software code contains information about the caller's identity or a reference to this information. The signalling protocol message issued by the caller's terminal is forwarded (box **42** of FIG. **4**) to the called party's terminal and the associated computer software code is accessed. This code is then executed on the processor **26** associated with the called party's terminal, provided that security provisions on the destination terminal allow this (box **43** of FIG. **4**). The executed code controls the destination terminal such that it displays the identity of the caller. For example, by playing a sample of the caller's voice or by displaying the caller's name on a visual display.

By using this method, the caller's identity is correct no matter which terminal the caller uses and it is not necessary to make use of CLID information.

The example described above of allowing a caller to control a destination terminal in order to display information about the caller's identity is only one embodiment of the present invention. More generally, the invention provides a way for callers to control a destination terminal by selecting computer software code for association with SIP messages, or any other suitable signalling protocol messages. This control of the destination terminal is of course subject to any security and access restriction arrangements that are set-up on the destination terminal. The default situation is that the destination terminal is controlled by its associated signalling protocol client and associated processor. Thus the caller does not have absolute control over the destination terminal except in cases where the security and access restrictions allow this.

A calling party (or other user) is able to select or create the computer software that is to be associated with the signalling protocol messages using a user interface such as a graphical user interface (GUI). Once this code is selected or created it is stored in a location that is accessible by the calling party's signalling protocol client. This location could be at the terminal itself, at a gateway from which the terminal subtends or at any other suitable location. In addition, rules or other criteria are stored which specify when particular pieces of the stored computer software are to be associated with SIP or other signalling protocol messages.

FIG. **3** is a flow diagram for a method of controlling a destination terminal from an originating terminal. Computer software code is first associated with a signalling protocol message (box **301**) and then that signalling protocol message is forwarded to a destination terminal (see box **302** of FIG. **3**). The computer software code is then executed on a processor associated with the destination terminal in order to control the destination terminal (box **303** of FIG. **3**).

The computer software code may be associated with the signalling protocol message in any suitable manner, for example, by adding the code to the message or adding a reference to the location of the code to the message. Any suitable signalling protocol messages may be used, such as session initiation protocol (SIP) messages. Appendix A gives more details about this.

Several different examples of ways in which the destination terminal is controlled are now described.

In one example, the caller is able to provide information about the priority of the call. In the past this has not been possible for conventional public switched telephone network systems where the CLID and ringing tone are all that is available to alert the called party to the call request. Answering machines can be used but in that case the called party must be available to listen to incoming calls and answer these if they are urgent.

FIG. **5** is a flow diagram of a method of controlling a destination terminal using information about the priority of the call. The calling party processor associates computer software code which contains information about the priority of the call (or a reference to the location of this information) with one or more signalling protocol messages (box **50** of FIG. **5**). When the signalling protocol message is received by the destination terminal, the code is executed as described above (box **51** of FIG. **5**) and this causes the priority information to be displayed and/or to affect the behaviour of the destination terminal (box **52** of FIG. **5**). For example, if the priority of the call is very urgent, then the code may cause the destination terminal to re-direct the call to an associated mobile telephone. As mentioned above this is subject to access and security restrictions. For example, the called party may have set up a database containing the identities of callers who should be given access at all times, those to whom access is to be denied and those to whom access should be given only during certain time periods. In this case, information about the caller's identity is obtained and used to determine which access levels are to be given.

In some situations, the destination terminal is engaged. In this case the computer software code associated with the signalling protocol message may be arranged to cause the destination terminal to be cleared (subject to security and access restrictions). For example, the caller may be trying to reach a family member urgently. The called party has previously stored the names of people who are allowed to cause the called party's terminal to be cleared of an "in progress" call. The called party may also have set up a password system whereby the caller must provide the password before being able to clear "in progress" calls. In this case, the computer software code sent by the caller with the signalling protocol message contains the password and/or name of the caller.

This process is illustrated in more detail in FIG. **6**. In the situation when the destination terminal is engaged (box **60** of FIG. **6**) the calling party processor associates computer software code, containing information about the caller's identity and code for clearing the "in progress" call from the destination terminal, with signalling protocol messages (box **61** of FIG. **6**). These messages are forwarded to the called party and the called party processor accesses the information about the identity of the caller (box **62** of FIG. **6**). The called party processor checks the identity of the caller against an access restriction database (previously set up by the called party). If access is granted to the particular caller, then the software code is executed in order to clear the "in progress" call from the destination terminal (box **63** of FIG. **6**).

The called party may also configure the signal processing client associated with its terminal such that "in progress" calls can only be cleared under certain circumstances. For example, when the "in progress" call is to an internet service provider or to one of a list of pre-specified destinations. In this way the called party is able to specify things like "If I am using the internet I am happy to allow family members to shut down my internet connection in order that they can telephone me." The called party is able to set up the security and access restrictions by using a user interface to modify the signal protocol client and any other software which controls the processor associated with the destination terminal.

The calling party is also able to select or create appropriate computer software code such that the configuration of the destination terminal is checked and taken into account before taking further action. For example, the number of rings at the destination terminal may be set to 3 before the call is diverted to a voice mail system. A caller may know that the called party only sets this number of rings to three when he or she is resting. In that case, the caller may prefer not to disturb the called party at all. The caller is then able to arrange the computer software code associated with the signalling protocol message such that it checks the "number of rings before divert to voice mail" setting at the destination terminal before proceeding with the call.

Known telephone systems, such as those in North America, have a facility whereby the calling party is able to block information about the CLID from the called party. This facility is often used by mobile phone callers who wish to prevent others from obtaining their mobile phone number. This is because they wish to prevent others from making calls to their mobile telephone which incur cost to the mobile phone owner. However, many non-mobile phone users have made use of the blocking facility, for example, sales people who wish to hide their identity in order that people will answer their calls. This has led to the creation of a service by which users are able to "block the blocker"; that is, users are able to block calls from any party who has blocked information about their identity from being made available to the called party. This "block the blocker" facility can be problematic in some circumstances. For example, consider a mobile phone user who has made use of the blocking facility. If that mobile phone user makes a call to a family member that family member is unaware of the identity of the caller. Suppose that the family member has implemented the "block the blocker" function on his or her terminal. In that case the user's call to the family member is blocked, even though that call may be extremely urgent. By making use of the present invention this problem is overcome. The user is able to control the family member's terminal in order to override the "block the blocker" function. For example, the caller sends signalling protocol messages containing a password which the called party receives and checks against pre-specified security criteria. If security clearance is obtained, software code associated with the signalling protocol messages ensures that the "block the blocker" function on the destination terminal is over-ridden.

In another example, the caller is able to control the destination terminal to give preferred handling to the call. For example, the caller is able to control the destination terminal such that the call is directed straight to a voice mail service or straight to the called party's mobile telephone. Prior art systems which allow a user to call a voice mail system directly (rather than being diverted to the voice mail system from the destination terminal) are difficult to use. Typically the caller must dial the number to connect to the voice mail system and then enter details about who is being called. This is time consuming and complex. By using the present invention this problem is avoided because a call with the called party is actually established unlike the prior art situation where a call is established directly with the voice mail system.

FIG. **7** is a flow diagram of a method for controlling a destination terminal in order that the call is directed straight to a voice mail system. The calling party processor associates computer software code with one or more signalling protocol messages (box **70**, FIG. **7**). This computer software code is arranged to control the destination terminal such that the call is directed straight to the voice mail system rather than causing the destination terminal to ring. The signalling protocol messages are forwarded to the called party (box **71** FIG. **7**) and the computer software code accessed and executed on the called party processor (subject to any security restrictions). A call is effectively established between the calling and called party at this stage, although the destination terminal does not ring. Because a call is effectively established, details about the called party are available. The computer software code accesses these details and controls the called party processor such that the call is directed to the called party's voice mail system. The information about the called party is also forwarded to that voice mail system so that the called party is not required to re-enter these details (box **72** of FIG. **7**).

In another example, a user is able to adjust the configuration of his or her terminal from a remote location. For example, that user acts as a calling party and calls his or her own terminal. Using the method described herein for controlling destination terminals, the user is then able to control his or her own terminal. For example, the user is able to adjust services such as "number of rings before call sent to voice mail" and other such terminating services from a remote location. This is achieved by associating appropriate computer software code with signalling protocol messages and forwarding these to the called party processor for execution.

FIG. **8** shows an originating terminal **80** in more detail. The originating terminal has:

- an input **81** arranged to access computer software code suitable for controlling said originating terminal;
- a processor **82** arranged to associate said computer software code in use with one or more signalling protocol messages; and
- an output **83** arranged to route said signalling protocol messages to the destination terminal in use.

It is not essential for the processor **82** to be integral with the originating terminal **80**. It is also possible for the processor to be physically separate from the originating terminal as long as communication between the processor and originating terminal is provided.

FIG. **9** shows a destination terminal **90** in more detail. The destination terminal **90** comprises:

- a signalling protocol client **91** arranged to receive one or more signalling protocol messages sent from an originating terminal;
- a processor **92** arranged to access any computer software code associated with received signalling protocol messages in use; and wherein said processor is arranged to execute such accessed computer software code such that the destination terminal is controlled.

As for the originating terminal, it is not essential for the processor **92** to be integral with the destination terminal **90**. The same applies for the signalling protocol client **91**.

However, communication between the processor **92** and the destination terminal **90** and between the signalling protocol client **91** and the destination terminal **90** must be provided.

FIG. **10** illustrates another embodiment of the present invention. A plurality of terminals **112** (such as telephone handsets) are connected to a connectionless communications network (such as an internet protocol communications network) via access nodes **111**. A database **113** is also provided which is accessible by each of the access nodes **112**. The database contains pre-specified information about the identity of each terminal **112** (for example, the calling line identifier (CLID)) and the name of a user associated with each terminal. When a caller initiates a call, information about the caller's identity is forwarded to the database **113** and used to update the database **113**. For example, the identity information is forwarded to the database **113** by being associated with a signalling protocol message that is forwarded to the database. Also, it is not essential for the database to be located separately from other components of the communications network. For example, the database may be incorporated into the access nodes **111**. The caller's identity is then accessed from the database **113** by a destination terminal and displayed at that destination terminal. By dynamically updating the database **113** in this way, the correct identity information is displayed no matter whether the user uses different terminals **112** or several users use the same terminal.

FIG. **11** is a flow diagram of a method of displaying information about the identity of a caller at a destination terminal comprising the steps of:

- providing a database comprising information about the identity of a plurality of originating terminals and a caller associated with each originating terminal (box **120**);
- initiating a call from an originating terminal to a destination terminal (box **121**);
- receiving information at the originating terminal about the identity of a caller and forwarding this information from the originating terminal to the database and updating the database with this information (box **122**); and
- accessing the identity of the caller associated with the originating terminal from the database and displaying that identity at the destination terminal (box **123**).

A range of applications are within the scope of the invention. These include situations in which it is required to control a destination terminal from an originating terminal. For example, to cause information about the identity of a caller to be displayed at the destination terminal. Another example involves providing information about the priority of a call and allowing the behaviour of the destination terminal to be adjusted in response to the call priority. As well as this, it is possible to clear an "in progress" call from an engaged destination terminal and to take into account configuration information on the destination terminal. Users are also able to adjust the configuration of terminating services on their terminals from a remote location.

APPENDIX A

A method of associating computer software code with signalling protocol messages such as Session Initiation Protocol (SIP) messages is now described by repeating some of the text from Nortel Network's earlier co-assigned U.S. patent application Ser. No. 09/520,853. However, it is not essential to use the improved SIP protocol described below.

Any suitable protocol and method for associating computer software code with signalling protocol messages may be used.

The term "SIP Client" is used to refer to a computer program that is arranged to control a communications network node such that it is able to send SIP messages such as SIP request messages. The computing platform that the SIP client runs on is referred to as a "host system". The communications network node either comprises the host system or is associated with the host system.

The term "Java virtual machine" is used to refer to a processor which is arranged to execute Java applets or Java byte code.

The term "mobile autonomous software agent" is used to refer to a computer program that is able to halt itself and move itself from a first processor to another processor that is connected to the first processor for example by a communications network. The computer program is referred to as being autonomous because it is able to "decide" where to move and what it will do independently of external requests. An example of a mobile autonomous software agent is a Java mobile agent. Details about Java mobile agents are given in the article, "Under the Hood: The architecture of aglets", by Bill Venners, JavaWorld April 1997 the contents of which are incorporated herein by reference.

By extending the SIP protocol increased functionality is provided. SIP messages are modified to carry computer software code such as Java applets or to carry an address such as an universal resource locator (URL) indicating where computer software code is stored. An application programming interface (API) is also defined which allows the computer software code to interact with a receiving host system. SIP clients are also modified in order that they execute the computer software code associated with the SIP messages before any other actions are taken as a result of receipt of the SIP message.

FIG. **12** shows a communications network **1001** comprising a plurality of communications network nodes **1010** each such node comprising:

- a SIP client **1011**;
- an input **1012** arranged to receive SIP messages which may be associated with computer software code; and
- a processor **1013** arranged such that in use, when a SIP message is received, any computer software code associated with that SIP message is executed by the processor. This processor is provided by the host system and may comprise a Java virtual machine or any other suitable processor. These communications network nodes are referred to as enhanced SIP nodes because they are arranged to allow the enhanced SIP process to work.

The communications network of FIG. **12** is used in conjunction with the method illustrated in FIG. **13** in order to implement the enhanced SIP process. FIG. **13** is a flow diagram of a method of communicating between a first and a second node in a communications network, each of said nodes comprising a SIP client, said method comprising the steps of:

- associating computer software code with a SIP message (box **1020** in FIG. **13**);
- sending the SIP message from the first SIP client associated with the first node to the second SIP client associated with the second node (box **1021** in FIG. **13**); and
- executing the computer software using the second node (box **1022** in FIG. **13**).

For example, FIG. **14** illustrates an example of how a plurality of enhanced SIP clients **1030**, **1031**, **1032**, **1033**, **1041** interact. Each SIP client is supported on a communications network node (not shown). SIP client A **1030** is connected to SIP client B **1031** via a communications link **1034** and SIP client B **1031** is connected to both SIP client C **1032** and SIP client D **1033** via communications links **1034**. SIP client B **1031** has a host system **1035** which comprises a Java virtual machine. SIP client D **1033** is also connected to SIP client E via a communications link. SIP client D and has a host system **1039** which comprises a Java mobile agent virtual machine and SIP client E **1041** also has a host system **1041** which comprises a Java mobile agent virtual machine **1042**.

Using the enhanced SIP protocol, computer software code such as Java applets are associated with a SIP message **1036**. That is, the computer software code may be added to the SIP message body itself or may be stored separately and an address of the storage location added to the SIP message. It is not essential to use Java applets or Java mobile agents; any other suitable computer software code may be used. The message **1036** is sent from SIP client A **1030** to SIP client B **1031**. SIP client B detects the presence of the Java applets (or other computer software code) associated with the SIP message **1036** and executes these Java applets using its Java virtual machine **1035** (or other type of host processor).

Any suitable method of detecting the presence of computer software code associated with the SIP message **1036** may be used. For example, an indicator may be placed in the header of the SIP message **1036** and the SIP client **1031** arranged to detect that indicator and associate it with the presence of computer software code. An example of such an indicator in a SIP message is described in more detail below.

By executing the Java applets, two new SIP messages **1037**, **1038** are created one of which **1037** contains a Java mobile agent and the other which does not. This is just one example of a something that the computer software code associated with the SIP message could do. For example, the computer software code could also be arranged to modify existing SIP messages, delete existing SIP messages, generate SIP messages, receive SIP messages or to control the SIP client and/or the host processor to perform any other suitable function. The computer software code is arranged to interact with the host processor via an API as described below. Security restrictions may be enforced by the SIP client and or host system in order to limit the actions that any software code associated with a SIP message is able to effect. More detail about these security restrictions is given below.

The executed Java applets then cause SIP client B **1031** to send one of the created messages **1037** to SIP client D **1033** and the other **1038** to SIP client C **1032**. The message **1037** sent to SIP client D contains a Java mobile agent (or other computer software code or an address of computer software code). If SIP client D has the capability to execute the Java mobile agent contained in message **1037** then SIP client D does so. However, if SIP client D does not have this capability, for example, if SIP client D has no Java mobile agent virtual machine, then SIP client D simply follows the standard SIP procedure for unsupported require extensions. This involves returning an error message to SIP client B, indicating that the Java applet in message **1037** was not executed.

In the meantime, SIP message **1038** which is not associated with any computer software code, is sent to SIP client C **1032** and any SIP process associated with that message **1038** is carried out following the standard SIP protocol.

In this example, SIP client D does have an associated Java mobile agent virtual machine **1039** and so when message **1037** arrives, the Java mobile agent in message **1037** begins to execute on this processor. At some point in the execution, the Java mobile agent suspends itself and includes itself in SIP message **1040** which is sent to SIP client E. This is one example of a process that may occur by incorporating a Java mobile agent into a SIP message.

In the enhanced SIP protocol described herein, standard SIP messages are modified by associating computer software code with them as described above. For example, one or more Java applets or Java mobile agents are stored in a multipart MIME section in the body of a SIP message or a URL indicating where the Java applets or Java mobile agents are stored is added to the SIP message.

In some examples, an indicator is added to the SIP message header, in order to indicate that computer software code is associated with that SIP message. For example, a "Require request-header" is used to indicate that Java enhanced SIP must be supported to process a SIP message that is associated with Java applets or Java byte code. This require request header is the same as the header for a standard SIP message except that the content type field in the entity header is used to indicate that the content type is a Java applet or the URL of a Java applet which must be retrieved. Also, the require field of the request-header is used to specify that Java enhanced SIP must be supported to process the message concerned.

FIG. **15** illustrates the structure of a standard SIP message and shows how this structure is used in the improved SIP protocol described herein. The structure of a standard SIP message is illustrated at **1040** in FIG. **15**. Thus a standard SIP message comprises a general-header, a request-header, an entity header, a CRLF and a message body. The structure of a general-header is shown at **1041** in FIG. **15** and similarly the structures of each of an entity header **1042**, request header **1043** and response header **1044** are shown. In order to indicate that the improved SIP protocol described herein is being used markers or tags are included in the SIP message in any suitable location. For example, the content-type field of an entity header may be used to indicate that the content type is a Java applet or the URL of a location of a Java applet. Similarly, the content-type field of an entity header may be used to indicate that the content type is a Java mobile agent or the URL of a location of a Java mobile agent. Also, the require field of a request header may be used to indicate that Java enhanced SIP must be supported to process the message concerned. However, it is not essential to use the content-type field or the require field for this purpose. Any other suitable field(s) may be used.

FIG. **16** shows an example of an INVITE message according to the improved SIP protocol described herein. The content type field contains the words "multipart/mixed" which indicates that the INVITE message body is in the form of a MIME multipart message which contains one or more Java applets or Java mobile agents. The require field contains the words "org.ieff.sip.java-enhanced-sip" which indicate that the improved SIP protocol must be used to process this message. Part of the body of the INVITE message containing the Java applet(s) or Java mobile agents is shown **1050**.

The SIP clients used to implement the improved SIP protocol are the same as standard SIP clients except that they are arranged to do the following things:

- Detect improved SIP messages which are associated with computer software code. For example, this may be done by arranging the SIP client to recognise the

presence of the words "org.ietf.sip.java-enhanced-sip" or "org.ieff.sip.java-mobile-agent-enhanced-sip" in the SIP message header.

If an improved SIP message is received and detected, the software code associated with that SIP message is accessed by the SIP client and executed on the SIP client's host processor. Preferably, this execution is carried out immediately, before processing the SIP message any further. For example, if a content type field in a SIP header indicates that a URL for a Java applet is present then the SIP client must immediately get the applet from the URL and execute the applet on a Java virtual machine associated with the SIP client. If the SIP client does not execute the software code then it is preferably arranged to respond by returning status code **420** (bad extension) and by listing org.ieff.sip-.java-enhanced-sip in an unsupported header. The SIP client may not execute the software code if it is unable to do so, for example, if no Java virtual machine is available, or if the SIP client decides not to do this, for example, for security reasons.

Match incoming SIP messages to patterns and in the event of a match "wake up" any waiting computer software code. This is described in more detail below.

The SIP client's host processor is modified as compared to a standard SIP client's host processor in that it must comprise a processor of a specific type. For example, a Java virtual machine in the case that Java applets are associated with the improved SIP messages. In the case that Java mobile agents are used, a Java mobile agent virtual machine is required. Also, the SIP client's host processor has access to or comprises an API to allow the computer software code associated with the improved SIP messages to interact with the SIP client. For example, in the case that Java applets are used, the SIP client's host has access to a set of Java classes or applets that are defined in a Java enhanced SIP API. This API allows access into the SIP client to allow SIP messages to be built and sent subject to security restrictions. Using the API received Java applets or Java mobile agents are able to generate and receive SIP messages using the receiving SIP client.

Passing of control between the computer software code associated with improved SIP messages and the SIP client concerned.

In the case that standard SIP messages are used, these are processed by SIP clients in the standard way and control remains with the SIP clients. However, in the improved SIP case described herein, any computer software code associated with a SIP message takes precedence over other standard SIP processes associated with the SIP message or with any other SIP messages received by a SIP client during processing of the computer software code.

For example, the computer software code associated with a SIP message can be arranged to initiate a SIP session and to wait for a SIP response before proceeding. During this waiting period, control remains with the computer software code. The computer software code is able to specify that it will go to sleep and wait for the next SIP message which matches a particular patter. In that case, the SIP client does no other actions during the sleep period. Alternatively, the computer software code can deal with any other incoming SIP messages itself during the sleep period. Thus control does not pass back to the SIP client until the computer software code wants it to even if SIP messages from other sessions are arriving.

Application Programming Interface (API)

As described above an API is specified in order that the computer software code associated with improved SIP messages is able to affect the SIP client. For example, this API allows a received Java applet or Java mobile agent access to the SIP messaging functions on the SIP client.

Examples of methods that the API supports comprise:

SendSIPMessage—sends a SIP message and establishes a context for the Session if one does not already exist. The invoker (which is the piece of software code which called this function) can indicate if it wants the message to be part of an existing Session. For example, the invoker could be a Java applet or Java mobile agent.

ReceiveSIPMessage—retrieves a SIP message from the Client's input buffer on a first in first out (FIFO) basis.

ReceivedMessageSummary—returns a summary of any received messages in the client's input buffer along with a count of messages received. If the client does not support buffering of input messages this is indicated.

QueryCapabilities—returns the capabilities of the Client. These include the ability to buffer incoming messages and the buffer size.

Querystatus—returns the status of any sessions the client is currently involved in.

MatchMessageAndWake—checks incoming messages against a particular pattern and if they match wakes up the indicated applet or Java mobile agent and passes the messages directly to the indicated applet.

ProcessMessage—sends a message to the Client and passes control to the client for the message to be processed as in standard SIP. For example, this can be used after an applet or Java mobile agent has looked at the message or altered it in some way and then wants to pass the message back to the client to be processed as in standard SIP.

ProcessMessageAndReturn—as for ProcessMessage except that control is passed back to the invoker after the message has been processed.

ProcessFromBufferAndReturn—processes the next message on the INPUT buffer as in standard SIP within the client and then returns control to the invoking applet or Java mobile agent.

Changes to SIP Proxy and SIP Server Behaviour

Following standard SIP as defined in "Request for comments (RFC) 2543 SIP: Session Initiation Protocol", SIP proxy and redirect servers must ignore features that are not understood. That is, if a SIP proxy or redirect server is not arranged to understand the improved SIP messages described herein then it must ignore features of those messages that are not common to standard SIP. A SIP proxy server is a communications network node which communicates using the SIP protocol on behalf of other parties. A SIP redirect server is a communications network node which receives SIP messages and directs these to another communications network node. If a particular extension to the standard SIP protocol requires that intermediate devices support it, the fact that the extension is being used must be tagged in the proxy-require field as well (see section 6.28 of the SIP RFC mentioned above). Thus for the improved SIP described herein, an indicator is placed in the proxy-require field to specify that the improved SIP is being used.

Security

Preferably, security mechanisms are incorporated in to the improved SIP protocol although this is not essential. For example, a host system which supports a SIP client preferably comprises security mechanisms for controlling the

activity of software code such as Java applets or Java mobile agents received as a result of the improved SIP messages. These security mechanisms may be configured by a user or operator, for example, to always allow or prevent certain operations from being carried out by Java applets or Java mobile agents received from improved SIP messages. The user may datafill a matrix of SIP operations against security mechanism actions. It is also possible for the security mechanism to prompt the user to ask for permission to proceed with certain actions. The security mechanisms are put into effect by a security manager which takes the form of a computer software application located at each SIP client. Preferably, all the methods specified in the API are arranged to check with the security manager at the SIP client concerned before proceeding with the rest of that method. In the case that Java byte code, Java applets or Java mobile agents are used, then the security mechanisms are preferably designed to conform to the standard Java security practices.

An example of an algorithm for a security mechanism is:

Index the matrix for user defined security checks against that operation

Extract the method corresponding to the security action datafilled by the user

Execute that security mechanism method

If the result of the security mechanism method is "pass" then continue and call the SIP API method

Else display a security disallowed message and return without calling the SIP API method.

Actions that a user may datafill for a given SIP operation include:

Allow always

Disallow always

Allow conditional

Disallow conditional

Prompt y/n

Allow and display warning or info

An example of use of the improved SIP protocol to create a service for automatically setting up multimedia conferences is now described.

Conferencing System

Using the improved SIP protocol a conferencing service is created whereby a single chairperson is able to set up the conference by sending out SIP INVITE messages. The method is suitable for multimedia conferences. The INVITE messages are associated with computer software code which executes on the host machines of invited attendees to set up the conference call. This greatly simplifies the process of setting up a conference call such as a multimedia conference call.

For example, the computer software code associated with the improved SIP INVITE messages can be arranged to set up connections from each attendee's machine to several video sources and to an electronic whiteboard to be shared for the meeting. The computer software code can also be arranged to start up a web browser to a page relevant to the meeting on each attendee's machine. As well as this the computer software code is able to set up all the audio paths between all the parties with everyone but the chairman initially on mute. As well as this the computer software code is able to take into account different capabilities of individual attendee's host machines. For example, a particular attendee such as a mobile caller may only have audio capabilities whilst a full multimedia caller may have audio, video, data and web capabilities. In order that these capabilities are taken into account, attendee's indicate what their capabilities are in SIP messages as required.

The multimedia conferencing service is particularly advantageous from the attendee's point of view. All the attendee has to do is to accept the incoming call and SIP INVITE message and everything will be set up for them automatically. Alternatively, the attendee may call a conference number and receive a SIP message in reply which is associated with the required computer software code. The conference number may be the number of a particular user client or of a central conference service provider.

Preferably security mechanisms are used in the multimedia conferencing service as described above.

FIG. **17** is a flow diagram of a method where a central conference service system is used and where Java applets are associated with the improved SIP messages. The first stage involves a user who wants to join a conference call sending a SIP INVITE message to the conference service system from his or her terminal (box **1060** FIG. **17**). This call is received by the conference service system which then returns an acknowledgement message ACK back to the user's terminal (box **1061** FIG. **17**). This ACK message is associated with one or more Java applets which contain methods from the API discussed above. The user's SIP client receives the ACK message, accesses the associated Java applet(s) and runs these using its associated Java virtual machine (box **1062** FIG. **17**).

The Java applet(s) query the exact capabilities of the user's SIP client and host machine and taking these capabilities into account, initiate SIP sessions for any audio, video and data streams associated with the conference as appropriate given the capabilities (box **1063** of FIG. **17**). Depending on how the user has his or her security mechanisms set he or she may be prompted before the sessions are set up for the various media streams. When the Java applet(s) initiate the SIP sessions (box **1063** of FIG. **17**) they may also be arranged to set up these SIP sessions such that all the attendees except for a chairperson are on mute. This is particularly advantageous, because the chairperson is then easily able to announce the beginning of the meeting and to chair the meeting in an organised fashion.

The Java applets(s) may also be arranged to forward details of a web page from each attendee to a chairperson or to the conference service system. For example, a web page giving biographical details of each attendee may be forwarded to a chairperson who then makes these available to each other attendee. In a similar manner, digital photographs of each attendee may be forwarded to the chairperson by the Java applets. It is also possible for the Java applets to request a joining message from each attendee which is then forwarded to a chairperson automatically by the Java applets. This joining message may contain security requirements specific to each attendee.

Depending on the number of parties to the conference, a conferencing bridge facility may be used as is known in the art. Alternatively, a software based technique is used to connect the parties to the conference.

An example of an algorithm that is encoded in the Java applet(s) of the method described immediately above is:

Read the message that the Java applet was associated with to obtain the addresses for the various streams in the call

Query the capabilities of the SIP client

Query the capabilities of the host system

Based on the above information for each media type and application available on the conference call:

If this application and media type is supported on the SIP client, initiate a SIP session between the SIP client and the relevant SIP client for that media stream.

Initiate a SIP message to the central conference service system detailing the number and types of streams set up.

FIG. **18** is a flow diagram of a method of setting up a conference call between two or more parties, each party comprising a SIP client and a host processor, said method comprising the steps of:

associating computer software code with a SIP message (box **1070** of FIG. **18**);

sending the SIP message to each of the parties (box **1071** of FIG. **18**);

executing the computer software code at each of the host processors (box **1072** of FIG. **18**).

FIG. **12** also shows a system for automatically setting up a conference call between two or more parties **1010**, each party comprising a SIP client **1011** and a host processor **1013**, said system comprising:—a processor **1013** for associating computer software code with a SIP message and to send that SIP message to each of the parties **1010**; and wherein each of said host processors **1013** is arranged to execute the computer software code in use, when the SIP message is received.

In the case that a conferencing system is used, this system sends the SIP messages to each party as a result of request calls from those parties to the system. In the case that a chairperson sets up the call, then the chairperson sends the SIP messages to each party.

Hunt Group System

An example of the use of improved SIP with Java mobile agents is now described. In this example, a service is provided whereby an automated system calls several telephones within a defined group (such as a team in an office) until one of those telephones is answered. For example, the nodes of the communications network in FIG. **12** may each provide a telephone implemented by software in the SIP clients **1011**. Each telephone within the group **1001** comprises a SIP client **1011** and a host processor **1013** as illustrated in FIG. **12** and the telephones are connected to one another via a communications network **1001** as shown in FIG. **1012**. The host processors each comprise a Java mobile agent virtual machine.

A user, which may be an automated service or a human using a terminal connected to the communications network **1001**, telephones one of the telephones **1010** within the defined group. If the called telephone is not answered after a specified number of rings or an elapsed time, then software at the SIP client **1011** of the called telephone creates a Java mobile agent, associates this with a SIP message, and sends the SIP message to a predefined second SIP client. This second SIP client is one of the telephones within the defined group **1001**.

The second SIP client receives the SIP message which is associated with the Java mobile agent. The Java mobile agent then executes itself on the Java mobile agent virtual machine associated with the second SIP client. The Java mobile agent is arranged to apply ringing to the second telephone and queries the second telephone's identification details and sends these back to the original caller. If the caller is using a host processor that has a display system associated with it, then information about the call and the fact that it has been forwarded to the second telephone in the defined group is sent by the Java mobile agent to this display.

If the second SIP client does not answer after a specified number of rings or time then the second SIP client repeats the method that the first SIP client carried out as described above. However, the second SIP client incorporates information about the fact that the call has been forwarded again.

After the method has been repeated a pre-determined number of times and if the call is not answered, then the call is sent back to the first SIP client that was called. A display of the route taken and the fact that the call was not answered is made at the first SIP client if a display is available.

If the call is answered, information about the route taken and the identity of the answering SIP client is sent back to the caller which may be an automated service.

FIG. **21** shows a method of forwarding a call from a first SIP client to a second SIP client, each of said SIP clients being associated with a host processor, said method comprising the steps of:

receiving a call at the first SIP client and if that call is not answered then associating computer software code with a SIP message said computer software code being arranged to forward a call (box **10100** FIG. **21**);

sending the SIP message from the first SIP client to a specified second SIP client (box **10101** FIG. **21**); and

executing the computer software using the host processor associated with the second SIP client such that the call is forwarded to the second SIP client (box **10102** FIG. **21**).

Client test system

Another example of the use of Java mobile agents with improved SIP involves a test system for a pre-defined group of SIP clients. For example, the network of SIP clients shown in FIG. **12**. The SIP clients **1011** are connected to one another to form a communications network **1001** as illustrated in FIG. **21**. Each SIP client **1011** is associated with a host processor **1013** which comprises a Java mobile agent virtual machine.

A test system (for example, software located at one of the nodes **1010** in the communications network **1001**), which may be an automated software service, creates a Java mobile agent, associates this with a SIP message, and sends that SIP message to one of the SIP clients **1011** in the group. The Java mobile agent executes on the receiving SIP client and sets up one or more test sessions. The results of these test sessions are stored by the Java mobile agent in its private data, together with any other required information. The Java mobile agent then associates itself with another SIP message and arranges that this SIP message be sent to another SIP client in the group. When the SIP message reaches another SIP client the process of obtaining information is repeated so that more information is added to the Java mobile agent's private data. Another SIP message is used to send the Java mobile agent on to another SIP client and so on, until all the SIP clients in the group have been visited. Once all the SIP client's in the group have been visited by the Java mobile agent, this agent associates itself with a SIP message in order to be sent back to the originating SIP client. In this way the Java mobile agent is able to report the results of its tests to the originating SIP client. The Java mobile agent may also be arranged to initiate other actions to fix any faults that it finds as it finds them. FIG. **20** shows a method of testing members of a group of SIP clients each SIP client being associated with a host processor said method comprising the steps of:

associating computer software code suitable for said testing with a SIP message (box **1090** FIG. **20**);

sending the SIP message one of the SIP clients (box **1091** FIG. **20**);

executing the computer software at the host processor associated with that SIP client in order to obtain test results (box **1092** FIG. **20**); and

repeating steps (ii) to (iii) for each of the other SIP clients in the group (box **1093** FIG. **20**).

Upgrade or replacement of SIP clients

Consider a situation in which it is required to upgrade or replace SIP clients which support the improved version of SIP described herein. This may be carried out automatically as follows:

The software for the upgrade or new SIP client is associated with a SIP message, for example, by building the software into a Java applet and adding this applet to a SIP message. This SIP message is then sent to all the SIP clients which are to be upgraded or replaced. On receipt of the SIP message at a SIP client, the existing SIP client runs the software code in order to effect the upgrade or replacement. The extent to which the upgrade or replacement is effected depends on the security specifications and the type of SIP client. By using the improved SIP protocol in this way, upgrades or replacement of a plurality of SIP clients is achieved quickly and easily.

FIG. **19** shows a method of upgrading or replacing interconnected SIP clients each SIP client being associated with a host processor said method comprising the steps of:—

associating computer software code suitable for said upgrade or replacement with a SIP message (box **1080** FIG. **19**);

sending the SIP message to each of the SIP clients (box **1081** FIG. **19**); and

executing the computer software at each of the host processors (box **1082** FIG. **19**).

What is claimed is:

1. A method of remotely controlling a destination terminal from an originating terminal said destination terminal having an associated signalling protocol client and an associated processor comprising the steps of:

(i) storing computer software code in at least one signalling protocol message;

(ii) sending the signalling protocol message to the destination terminal from the originating terminal;

(iii) executing the computer software code using the processor associated with the destination terminal in order that the originating terminal controls the destination terminal.

2. A method as claimed in claim **1** wherein said step (iii) of executing further comprises activating a security means at the destination terminal and executing the computer software code depending on the activated security means.

3. A method as claimed in claim **1** wherein said computer software code is arranged to access information about the identity of a caller.

4. A method as claimed in claim **3** wherein said computer software code is further arranged to display the identity information at the destination terminal.

5. A method as claimed in claim **1** wherein said computer software code is arranged to access information about a priority level for a call associated with the signalling protocol message.

6. A method as claimed in claim **1** wherein said computer software code is arranged to detect whether the destination terminal is engaged, and if so to clear the destination terminal in order that it is able to accept an incoming call associated with the signalling protocol message.

7. A method as claimed in claim **1** wherein said computer software code is arranged to access information from the destination terminal about the configuration of that terminal.

8. A method as claimed in claim **7** wherein said computer software code is further arranged to control the destination terminal on the basis of accessed configuration information.

9. A method as claimed in claim **1** wherein said computer software code is arranged to modify the configuration of terminating services associated with the destination terminal.

10. A method as claimed in claim **1** wherein said computer software code is arranged to direct a call associated with the signalling protocol message to a voice mail system associated with a called party.

11. A method as claimed in claim **1** wherein said signalling protocol message is a session initiation protocol (SIP) message and wherein said computer software code is selected from: Java byte code, Java applets and mobile automated software agents.

12. A destination terminal comprising:—

(i) a signalling protocol client arranged to receive one or more signalling protocol messages sent from an originating terminal;

(ii) a processor arranged to access any computer software code stored in received signalling protocol messages in use; and wherein said processor is arranged to execute such accessed computer software code such that the destination terminal is controlled.

13. A destination terminal as claimed in claim **12** which further comprises stored security information and wherein said processor is arranged to check said security information before executing the accessed computer software code.

* * * * *

JS 44 (Rev. 11/04)

CIVIL COVER SHEET

The JS 44 civil cover sheet and the information contained herein neither replace nor supplement the filing and service of pleadings or other papers as required by law, except as provided by local rules of court. This form, approved by the Judicial Conference of the United States in September 1974, is required for the use of the Clerk of Court for the purpose of initiating the civil docket sheet. (SEE INSTRUCTIONS ON THE REVERSE OF THE FORM.)

I. (a) PLAINTIFFS

Vonage Holdings, Corp.

DEFENDANTS

Nortel Networks, Inc.; Nortel Networks, Ltd.

(b) County of Residence of First Listed Plaintiff
(EXCEPT IN U.S. PLAINTIFF CASES)

County of Residence of First Listed Defendant
(IN U.S. PLAINTIFF CASES ONLY)
NOTE: IN LAND CONDEMNATION CASES, USE THE LOCATION OF THE LAND INVOLVED.

(c) Attorney's (Firm Name, Address, and Telephone Number)

See attachment.

Attorneys (If Known)

II. BASIS OF JURISDICTION (Place an "X" in One Box Only)

- ☐ 1 U.S. Government Plaintiff
- ☒ 3 Federal Question (U.S. Government Not a Party)
- ☐ 2 U.S. Government Defendant
- ☐ 4 Diversity (Indicate Citizenship of Parties in Item III)

III. CITIZENSHIP OF PRINCIPAL PARTIES (Place an "X" in One Box for Plaintiff and One Box for Defendant)
(For Diversity Cases Only)

	PTF	DEF		PTF	DEF
Citizen of This State	☐ 1	☐ 1	Incorporated *or* Principal Place of Business In This State	☐ 4	☐ 4
Citizen of Another State	☐ 2	☐ 2	Incorporated *and* Principal Place of Business In Another State	☐ 5	☐ 5
Citizen or Subject of a Foreign Country	☐ 3	☐ 3	Foreign Nation	☐ 6	☐ 6

IV. NATURE OF SUIT (Place an "X" in One Box Only)

CONTRACT
- ☐ 110 Insurance
- ☐ 120 Marine
- ☐ 130 Miller Act
- ☐ 140 Negotiable Instrument
- ☐ 150 Recovery of Overpayment & Enforcement of Judgment
- ☐ 151 Medicare Act
- ☐ 152 Recovery of Defaulted Student Loans (Excl. Veterans)
- ☐ 153 Recovery of Overpayment of Veteran's Benefits
- ☐ 160 Stockholders' Suits
- ☐ 190 Other Contract
- ☐ 195 Contract Product Liability
- ☐ 196 Franchise

REAL PROPERTY
- ☐ 210 Land Condemnation
- ☐ 220 Foreclosure
- ☐ 230 Rent Lease & Ejectment
- ☐ 240 Torts to Land
- ☐ 245 Tort Product Liability
- ☐ 290 All Other Real Property

TORTS

PERSONAL INJURY
- ☐ 310 Airplane
- ☐ 315 Airplane Product Liability
- ☐ 320 Assault, Libel & Slander
- ☐ 330 Federal Employers' Liability
- ☐ 340 Marine
- ☐ 345 Marine Product Liability
- ☐ 350 Motor Vehicle
- ☐ 355 Motor Vehicle Product Liability
- ☐ 360 Other Personal Injury

PERSONAL INJURY
- ☐ 362 Personal Injury - Med. Malpractice
- ☐ 365 Personal Injury - Product Liability
- ☐ 368 Asbestos Personal Injury Product Liability

PERSONAL PROPERTY
- ☐ 370 Other Fraud
- ☐ 371 Truth in Lending
- ☐ 380 Other Personal Property Damage
- ☐ 385 Property Damage Product Liability

CIVIL RIGHTS
- ☐ 441 Voting
- ☐ 442 Employment
- ☐ 443 Housing/ Accommodations
- ☐ 444 Welfare
- ☐ 445 Amer. w/Disabilities - Employment
- ☐ 446 Amer. w/Disabilities - Other
- ☐ 440 Other Civil Rights

PRISONER PETITIONS
- ☐ 510 Motions to Vacate Sentence

Habeas Corpus:
- ☐ 530 General
- ☐ 535 Death Penalty
- ☐ 540 Mandamus & Other
- ☐ 550 Civil Rights
- ☐ 555 Prison Condition

FORFEITURE/PENALTY
- ☐ 610 Agriculture
- ☐ 620 Other Food & Drug
- ☐ 625 Drug Related Seizure of Property 21 USC 881
- ☐ 630 Liquor Laws
- ☐ 640 R.R. & Truck
- ☐ 650 Airline Regs.
- ☐ 660 Occupational Safety/Health
- ☐ 690 Other

LABOR
- ☐ 710 Fair Labor Standards Act
- ☐ 720 Labor/Mgmt. Relations
- ☐ 730 Labor/Mgmt.Reporting & Disclosure Act
- ☐ 740 Railway Labor Act
- ☐ 790 Other Labor Litigation
- ☐ 791 Empl. Ret. Inc. Security Act

BANKRUPTCY
- ☐ 422 Appeal 28 USC 158
- ☐ 423 Withdrawal 28 USC 157

PROPERTY RIGHTS
- ☐ 820 Copyrights
- ☒ 830 Patent
- ☐ 840 Trademark

SOCIAL SECURITY
- ☐ 861 HIA (1395ff)
- ☐ 862 Black Lung (923)
- ☐ 863 DIWC/DIWW (405(g))
- ☐ 864 SSID Title XVI
- ☐ 865 RSI (405(g))

FEDERAL TAX SUITS
- ☐ 870 Taxes (U.S. Plaintiff or Defendant)
- ☐ 871 IRS—Third Party 26 USC 7609

OTHER STATUTES
- ☐ 400 State Reapportionment
- ☐ 410 Antitrust
- ☐ 430 Banks and Banking
- ☐ 450 Commerce
- ☐ 460 Deportation
- ☐ 470 Racketeer Influenced and Corrupt Organizations
- ☐ 480 Consumer Credit
- ☐ 490 Cable/Sat TV
- ☐ 810 Selective Service
- ☐ 850 Securities/Commodities/ Exchange
- ☐ 875 Customer Challenge 12 USC 3410
- ☐ 890 Other Statutory Actions
- ☐ 891 Agricultural Acts
- ☐ 892 Economic Stabilization Act
- ☐ 893 Environmental Matters
- ☐ 894 Energy Allocation Act
- ☐ 895 Freedom of Information Act
- ☐ 900Appeal of Fee Determination Under Equal Access to Justice
- ☐ 950 Constitutionality of State Statutes

V. ORIGIN (Place an "X" in One Box Only)

☒ 1 Original Proceeding ☐ 2 Removed from State Court ☐ 3 Remanded from Appellate Court ☐ 4 Reinstated or Reopened ☐ 5 Transferred from another district (specify) ☐ 6 Multidistrict Litigation ☐ 7 Appeal to District Judge from Magistrate Judgment

VI. CAUSE OF ACTION

Cite the U.S. Civil Statute under which you are filing (**Do not cite jurisdictional statutes unless diversity**):
28 U.S.C. 2201

Brief description of cause:
Declaratory judgment of non-infringement / invalidity

VII. REQUESTED IN COMPLAINT:

☐ CHECK IF THIS IS A CLASS ACTION UNDER F.R.C.P. 23

DEMAND $ Declaratory Judgment

CHECK YES only if demanded in complaint:
JURY DEMAND: ☒ Yes ☐ No

VIII. RELATED CASE(S) IF ANY

(See instructions): JUDGE DOCKET NUMBER

DATE
August 17, 2007

SIGNATURE OF ATTORNEY OF RECORD
Steven J. Balick

FOR OFFICE USE ONLY

RECEIPT # AMOUNT APPLYING IFP JUDGE MAG. JUDGE

CIVIL COVER SHEET ATTACHMENT

§ I(a) – Attorneys (Firm Name, Address, and Telephone Number)

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UNITED STATES DISTRICT COURT

District of

NOTICE, CONSENT, AND ORDER OF REFERENCE — EXERCISE OF JURISDICTION BY A UNITED STATES MAGISTRATE JUDGE

Plaintiff
V.

Case Number:

Defendant

NOTICE OF AVAILABILITY OF A UNITED STATES MAGISTRATE JUDGE TO EXERCISE JURISDICTION

In accordance with the provisions of 28 U.S.C. §636(c), and Fed.R.Civ.P. 73, you are notified that a United States magistrate judge of this district court is available to conduct any or all proceedings in this case including a jury or nonjury trial, and to order the entry of a final judgment. Exercise of this jurisdiction by a magistrate judge is, however, permitted only if all parties voluntarily consent.

You may, without adverse substantive consequences, withhold your consent, but this will prevent the court's jurisdiction from being exercised by a magistrate judge. If any party withholds consent, the identity of the parties consenting or withholding consent will not be communicated to any magistrate judge or to the district judge to whom the case has been assigned.

An appeal from a judgment entered by a magistrate judge shall be taken directly to the United States court of appeals for this judicial circuit in the same manner as an appeal from any other judgment of this district court.

CONSENT TO THE EXERCISE OF JURISDICTION BY A UNITED STATES MAGISTRATE JUDGE

In accordance with provisions of 28 U.S.C. §636(c) and Fed.R.Civ.P. 73, the parties in this case consent to have a United States magistrate judge conduct any and all proceedings in this case, including the trial, order the entry of a final judgment, and conduct all post-judgment proceedings.

Party Represented	Signatures	Date

ORDER OF REFERENCE

IT IS ORDERED that this case be referred to ______________________________
United States Magistrate Judge, to conduct all proceedings and order the entry of judgment in accordance with 28 U.S.C. §636(c) and Fed.R.Civ.P. 73.

Date ______________ United States District Judge ______________

NOTE: RETURN THIS FORM TO THE CLERK OF THE COURT ONLY IF ALL PARTIES HAVE CONSENTED ON THIS FORM TO THE EXERCISE OF JURISDICTION BY A UNITED STATES MAGISTRATE JUDGE.